Advanced Network Technologies

Application layer

Transport layer

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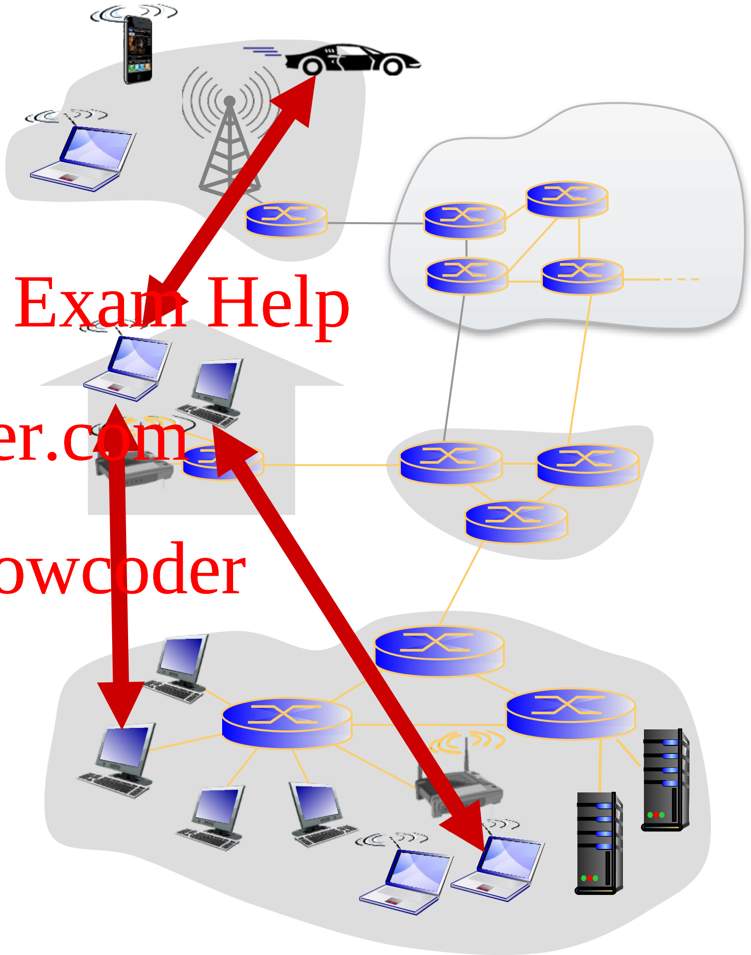
Peer-to-Peer

Pure peer-to-peer model architecture

- *no* always-on server
- arbitrary end systems directly communicate
- peers are intermittently connected and change IP addresses



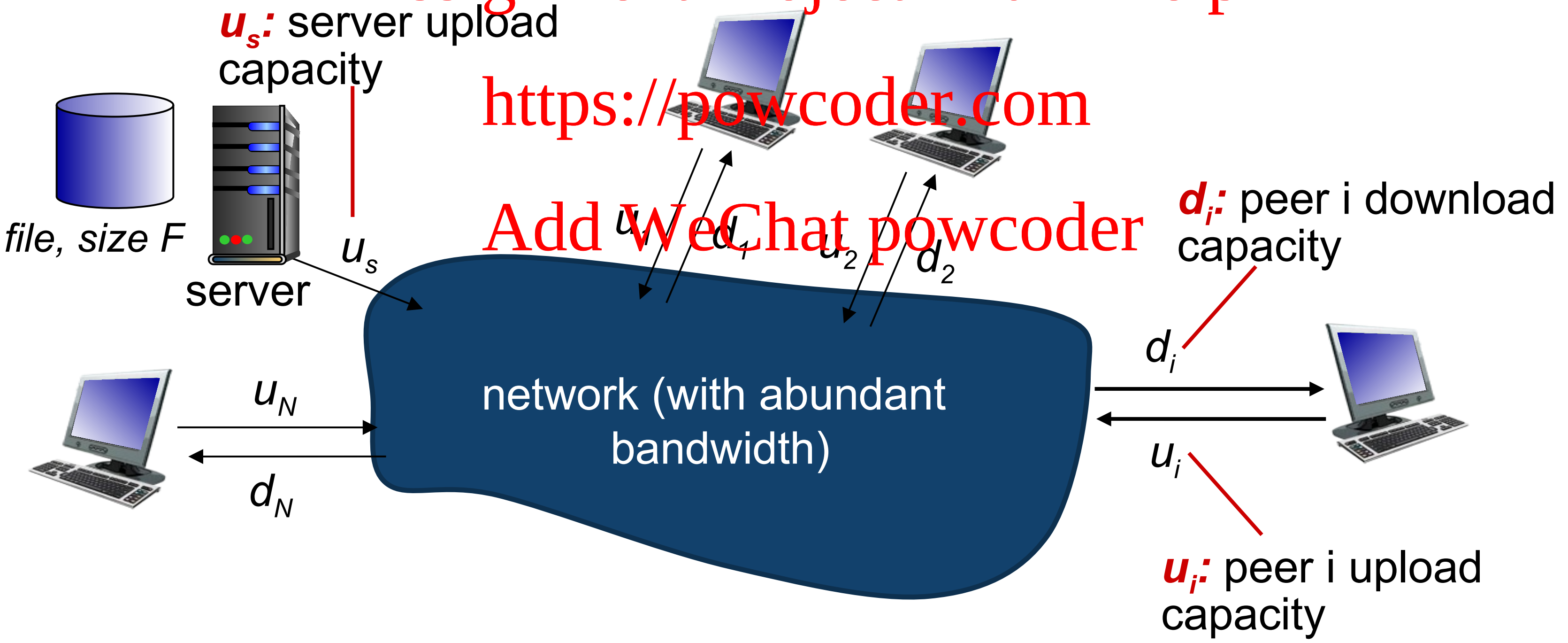
examples:

- file distribution (BitTorrent)
- Streaming (Zattoo, KanKan)
- VoIP (Skype)

File distribution: client-server vs. p2p

Question: how much time to distribute file (size *F*) from one server to *N* peers?

- peer upload/download capacity is limited resource



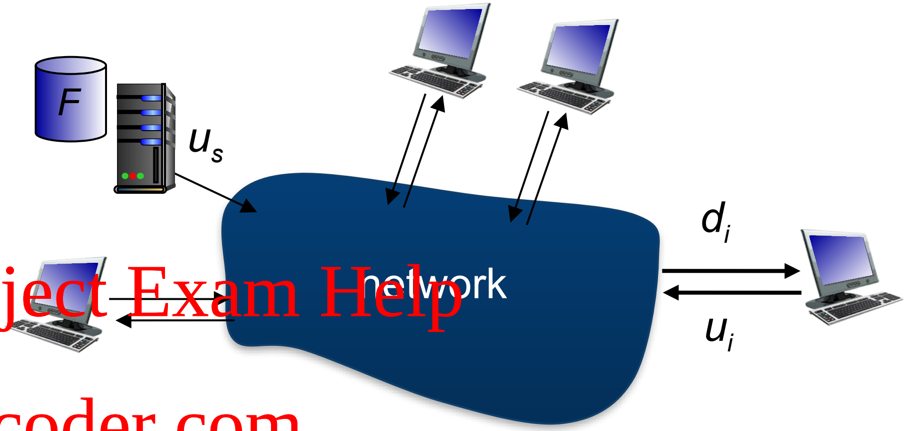
u_s: server upload capacity

d_i: peer i download capacity

u_i: peer i upload capacity

File distribution time: client-server

- *server transmission:* must sequentially send (upload) *N* file copies:
  - time to send one copy: F/u_s
  - time to send N copies: NF/u_s
- *client:* each client must download file copy
  - d_{min} = min client download rate
  - (worst case) client download time: F/d_{min}



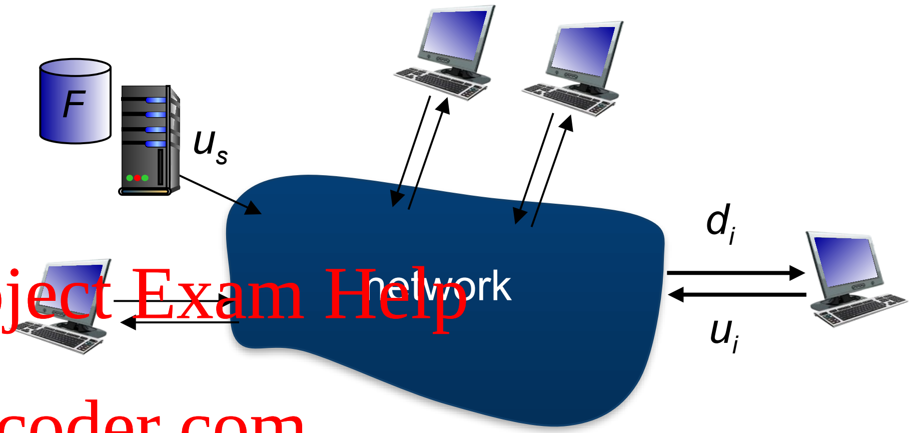
time to distribute F to N clients using client-server approach

$$D_{c\text{-}s} \geq max\{NF/u_{s,},F/d_{min}\}$$

increases linearly in N

File distribution time: p2p

- *server transmission:* must upload at least one copy
  - time to send one copy: F/u_s
    - *client:* each client must download file copy
      - client download time: F/d_{min}
    - *clients:* as aggregate must download *NF* bits = upload *NF* bits
      - Max upload rate $u_s + \Sigma u_i$
      - $NF/(u_s + \Sigma u_i)$



time to distribute F to N clients using P2P approach

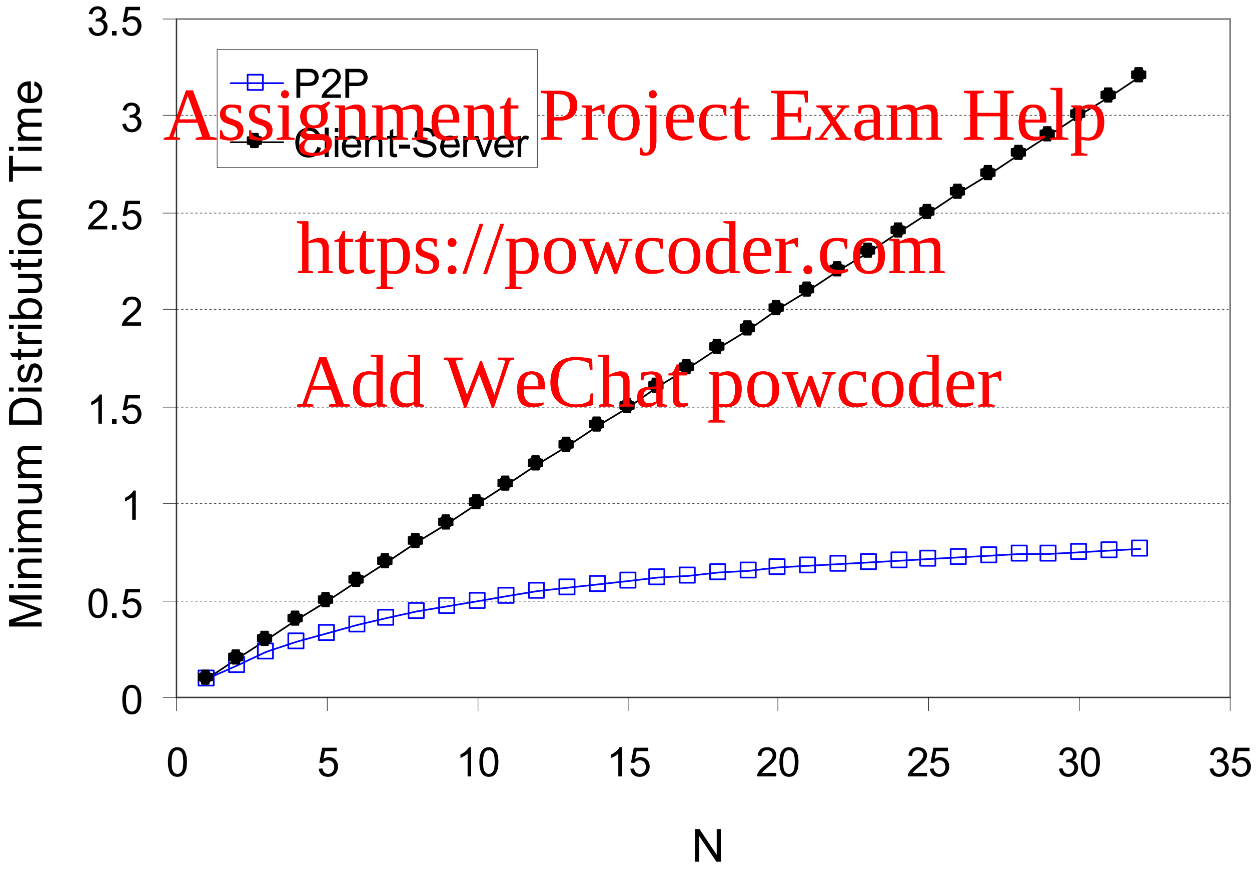
$$D_{P2P} \geq max\{F/u_{s,},F/d_{min,},NF/(u_s + \Sigma u_i)\}$$

increases linearly in *N* …

… but so does this, as each peer brings service capacity

Client-server vs. p2p

client upload rate = u, F/u = 1 hour, $u_s = 10u$, $d_{min} \geq u_s$



P2P file distribution: BitTorrent

BitTorrent, a file sharing application

- 20% of European internet traffic in 2012.
- Used for Linux distribution, software patches, distributing movies
- Goal: quickly replicate large files to large number of clients

- Web server hosts a .torrent file (w/ file length, hash, tracker's URL…)
- A tracker tracks downloaders/owners of a file
- Files are divided into chunks (256kB-1MB)
- Downloaders download chunks from themselves (and owners)
- Tit-for-tat: the more one shares (server), the faster it can download (client)

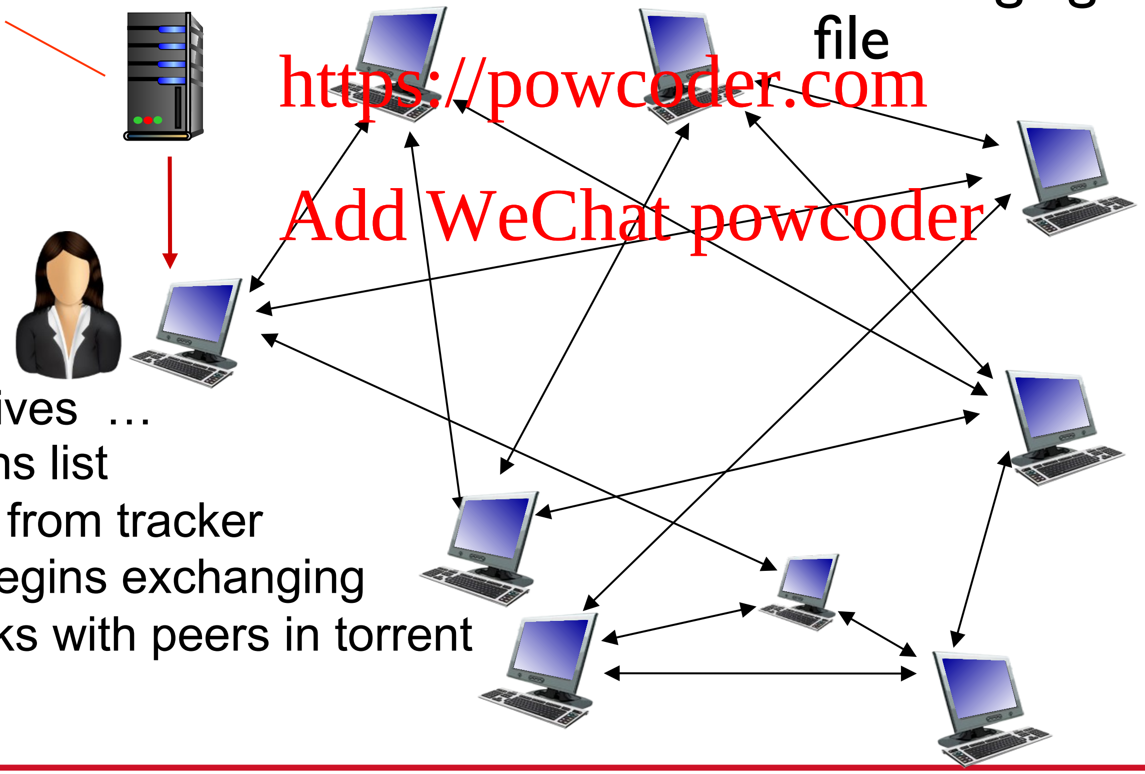
P2P file distribution: BitTorrent

› file divided into 256KB chunks

› peers in torrent send/receive file chunks

tracker: tracks peers participating in torrent

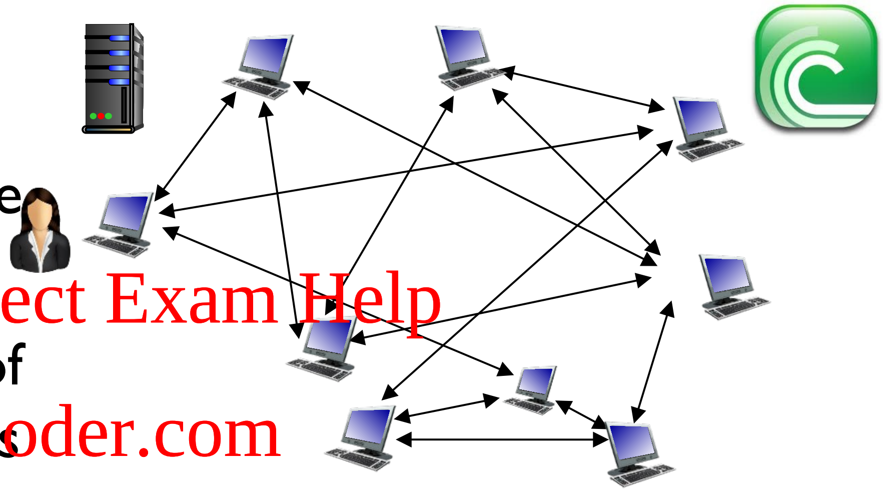
torrent: group of peers exchanging chunks of a file



Alice arrives …
… obtains list
of peers from tracker
… and begins exchanging
file chunks with peers in torrent

P2P file distribution: BitTorrent

- peer joining torrent:
  - has no chunks, but will accumulate them over time from other peers
  - registers with tracker to get list of peers, connects to subset of peers ("neighbors")



- while downloading, peer uploads chunks to other peers
- peer may change peers with whom it exchanges chunks
- *churn:* peers may come and go
- once peer has entire file, it may (selfishly) leave or (altruistically) remain in torrent

BitTorrent: requesting, sending file chunks

requesting chunks:

- at any given time, different peers have different subsets of file chunks
- periodically, Alice asks each peer for list of chunks that they have
- Alice requests missing chunks from peers, rarest first

sending chunks: tit-for-tat

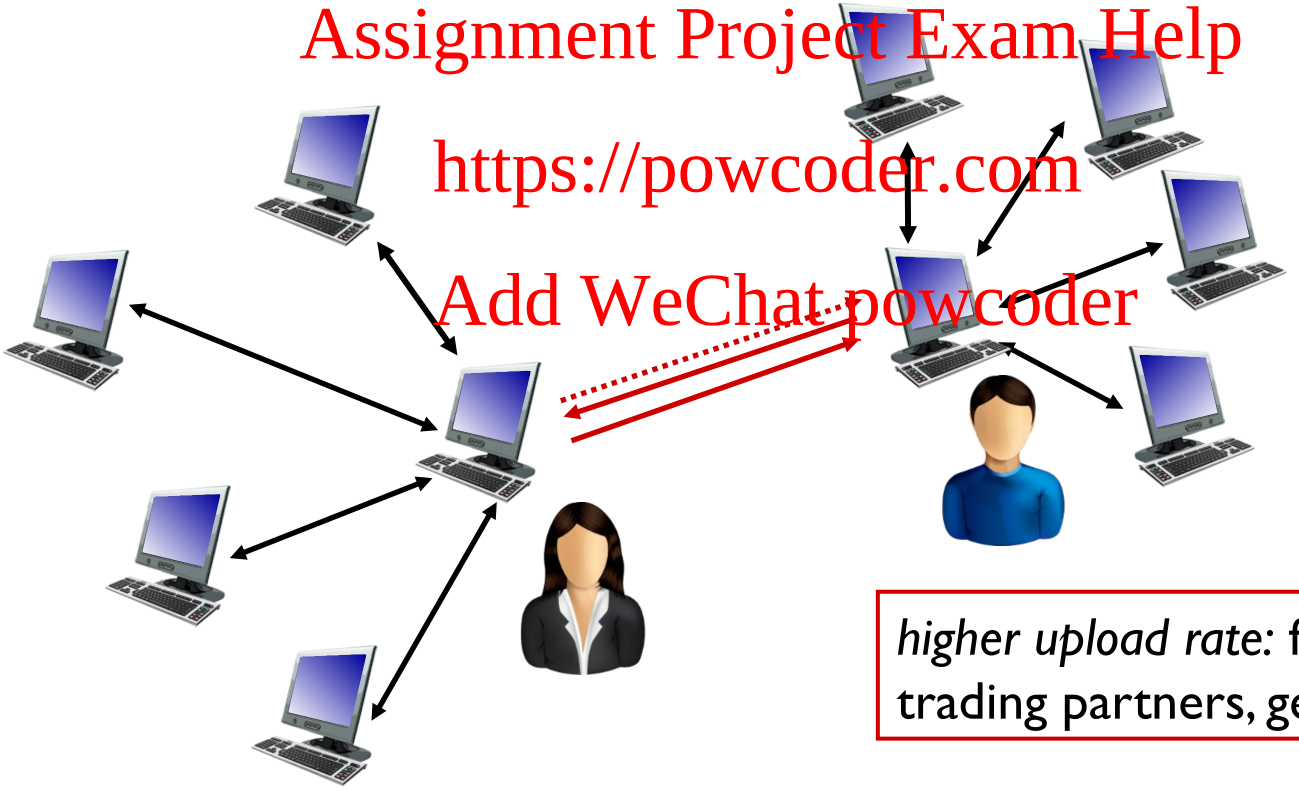
- Alice sends chunks to those four peers currently sending her chunks *at highest rate*
  - other peers are choked by Alice (do not receive chunks from her)
  - re-evaluate top 4 every10 secs
- every 30 secs: randomly select another peer, starts sending chunks
  - "optimistically unchoke" this peer
  - newly chosen peer may join top 4

BitTorrent: tit-for-tat

(1) Alice "optimistically unchokes" Bob

(2) Alice becomes one of Bob's top-four providers; Bob reciprocates

(3) Bob becomes one of Alice's top-four providers



higher upload rate: find better trading partners, get file faster !

Distributed Hash Table

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Distributed hash table (DHT)

- DHT: a *distributed P2P database*
- database has (key, value) pairs; examples:
  - key: social security number; value: human name
- distribute the (key, value) pairs over the many peers
- a peer queries DHT with key
  - DHT returns values that match the key
- peers can also insert (key, value) pairs

Distributed hash table (DHT)

- Assign the keys
- Lookup the keys

Assigning key to peer

- central issue:
  - assigning (key, value) pairs to peers.
- basic idea:
  - Key: generate an integer
  - Assign an integer ID to each peer
  - put (key,value) pair in the peer that is closest to the key

Assigning key to peer

- distance: assign integer identifier to each peer in range $[0,2^n-1]$ for some *n*.
  - each identifier represented by *n* bits.

- Each key to be an integer in same range $[0,2^n-1]$

- to get integer key, hash original key

  - A hash function is any function that can be used to map data of arbitrary size to data of fixed size (e.g., an integer in $[0,2^n-1]$).
  - e.g., 15 = hash(“Led Zeppelin IV”)
  - this is why its is referred to as a *distributed “hash” table*

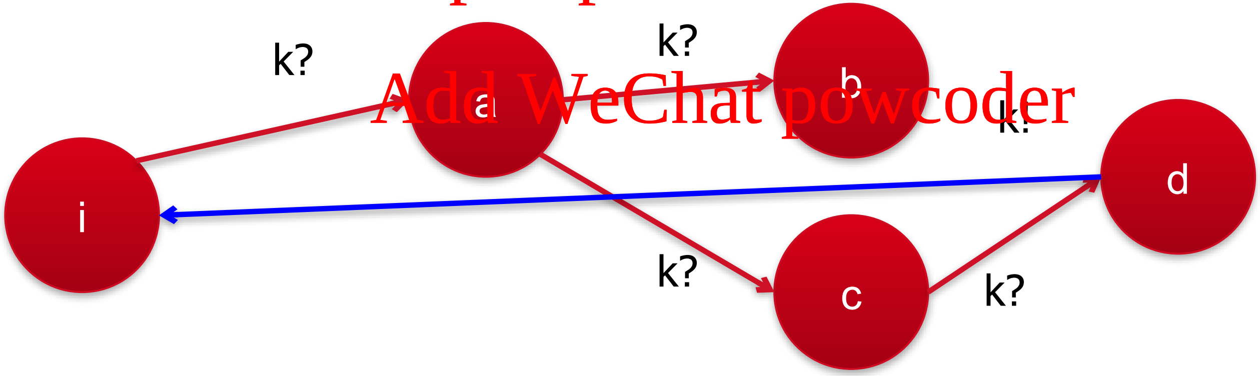
Assigning key to peer

- rule: assign key to the peer that has the *closest* ID.
- Here: closest is the *immediate successor* of the key.
- e.g., n = 4; peers: 1, 3, 4, 5, 8, 10, 12, 14;
  - key = 13, then successor peer = 14
  - key = 15, then successor peer = 1

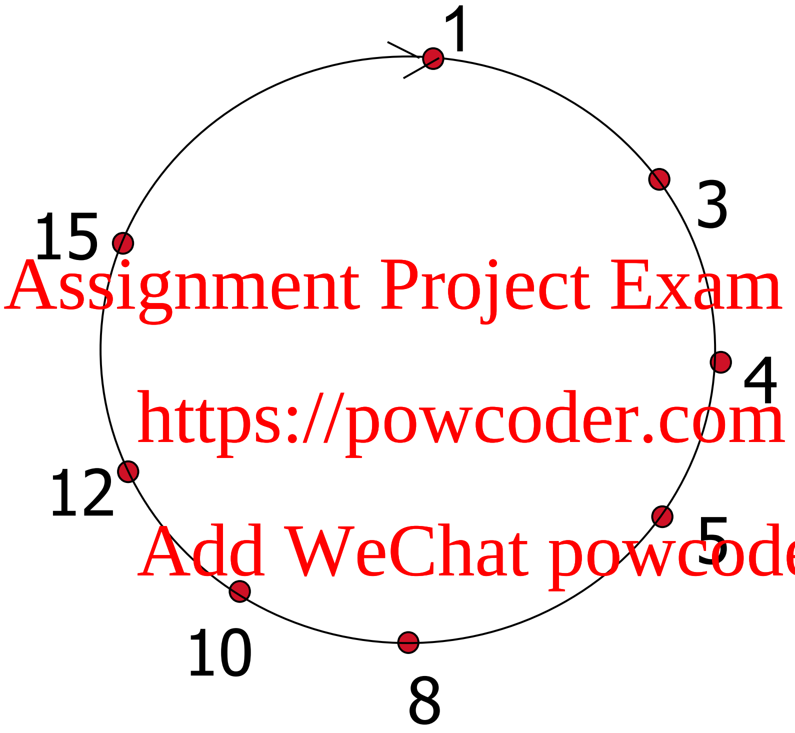
Lookup the keys

Goal: to provide a distributed lookup service returning the host that owns the key

› Given a key, find the host that owns the key



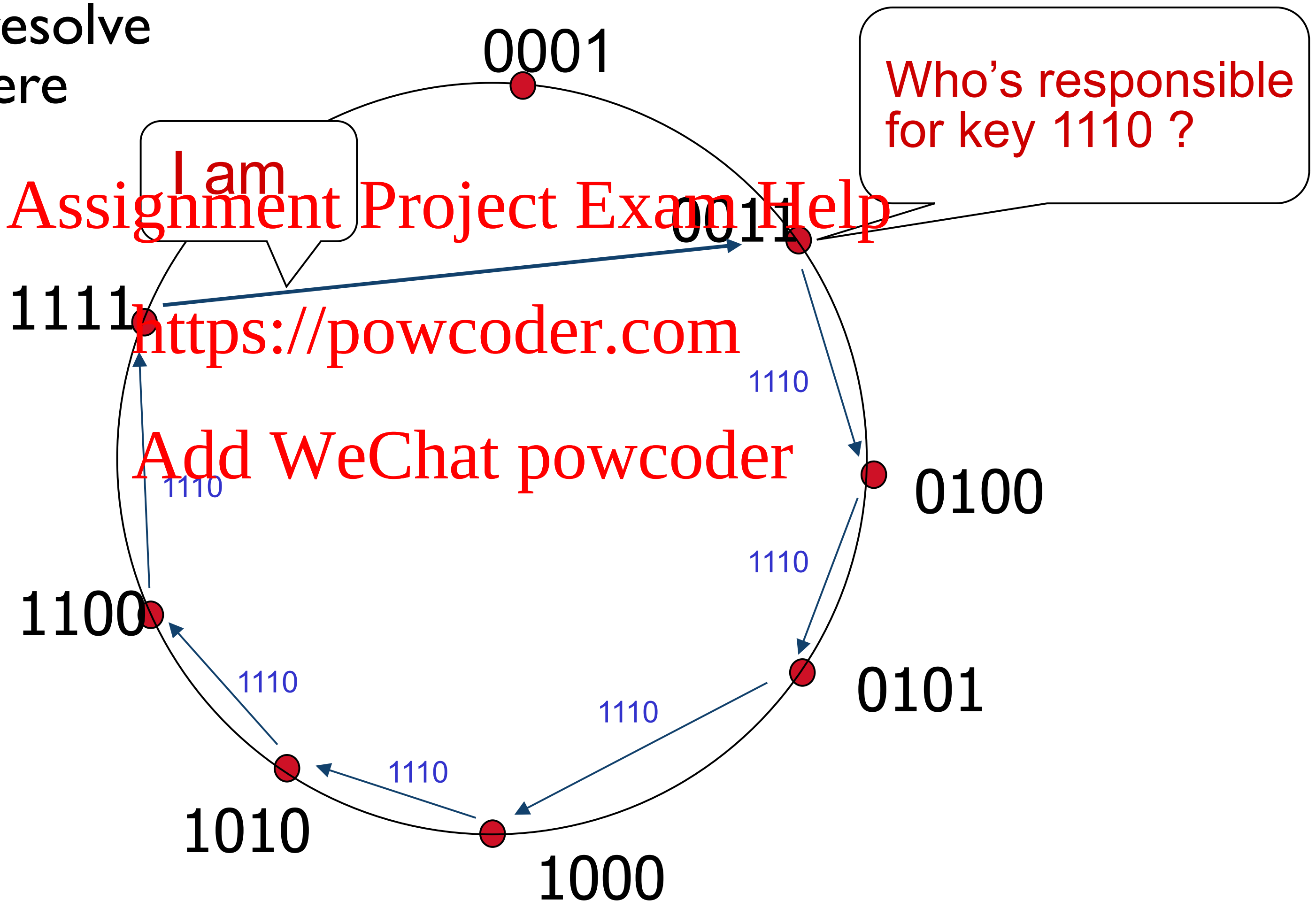
Circular DHT



- each peer *only* aware of immediate successor.

Circular DHT (con't)

O(*N*) messages on average to resolve query, when there are *N* peers



key 1110 is stored at node 1111

Define closest as closest successor

Circular DHT (con’t)

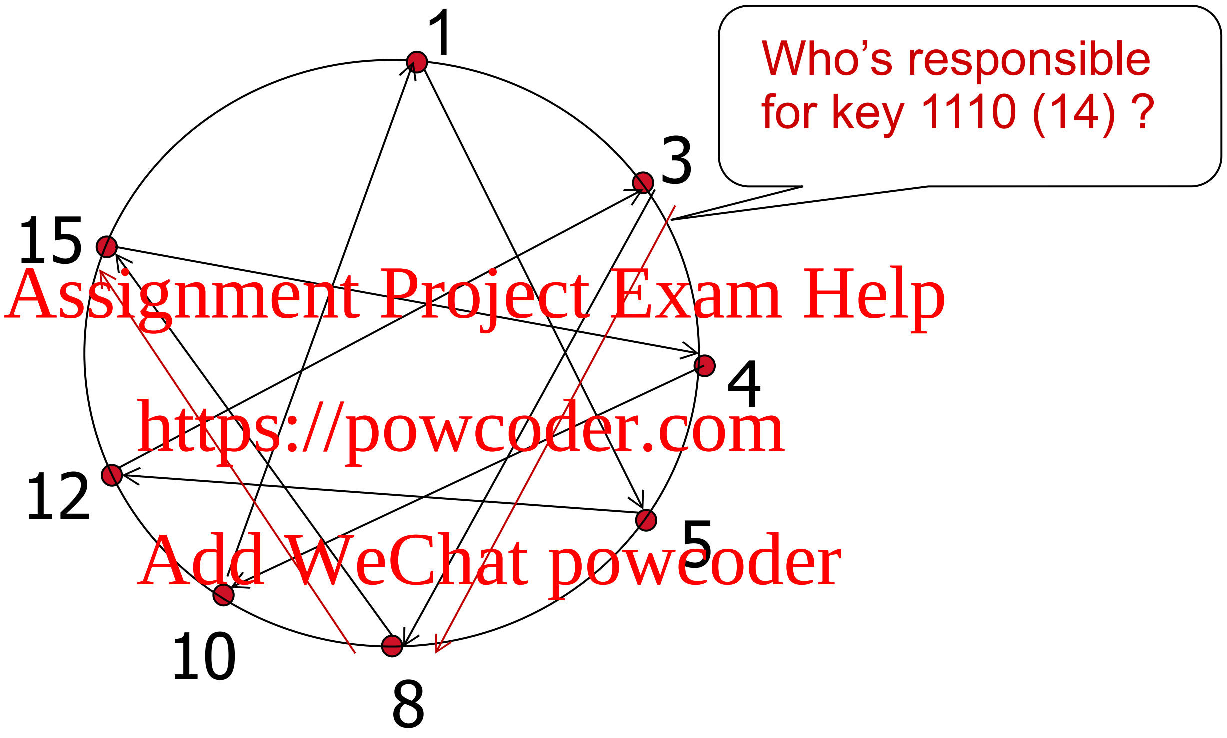
Example: Chord is an example of a Distributed Hash Table (DHT)

As a node:

- I have a successor peer
- I have a predecessor peer
- I have some shortcuts to other nodes to speedup delivery of requests

- Chord: A scalable peer-to-peer lookup service for internet applications. Stoica et al. *SIGCOMM* 2001.

Circular DHT (con't)



Who's responsible for key 1110 (14) ?

- each peer keeps track of predecessor, successor, short cuts.
- reduced from 6 to 2 messages.
- possible to design shortcuts so *O(log N)* neighbors, *O(log N)* messages in query

Transport Layer

Transport Layer

our goals:

- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control

- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion control

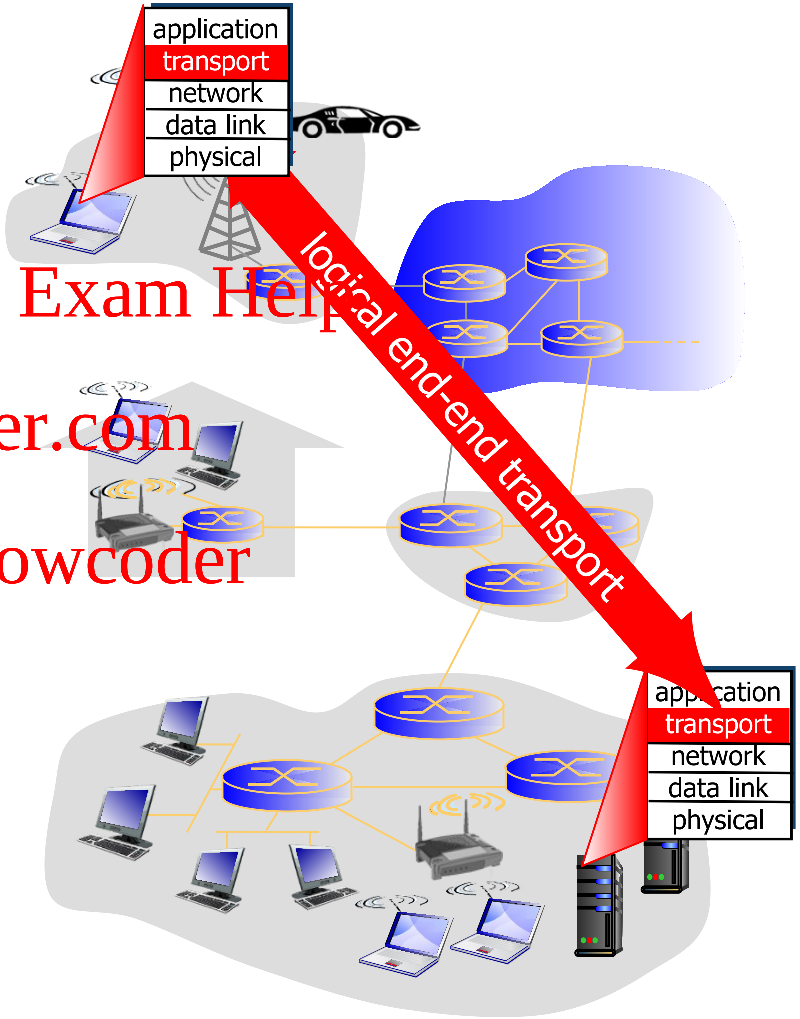
Outline

- Transport-layer services
- Multiplexing/demultiplexing
- Connectionless transport (UDP)
- Principles of reliable data transfer
- TCP protocol

Transport Services

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into *segments*, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



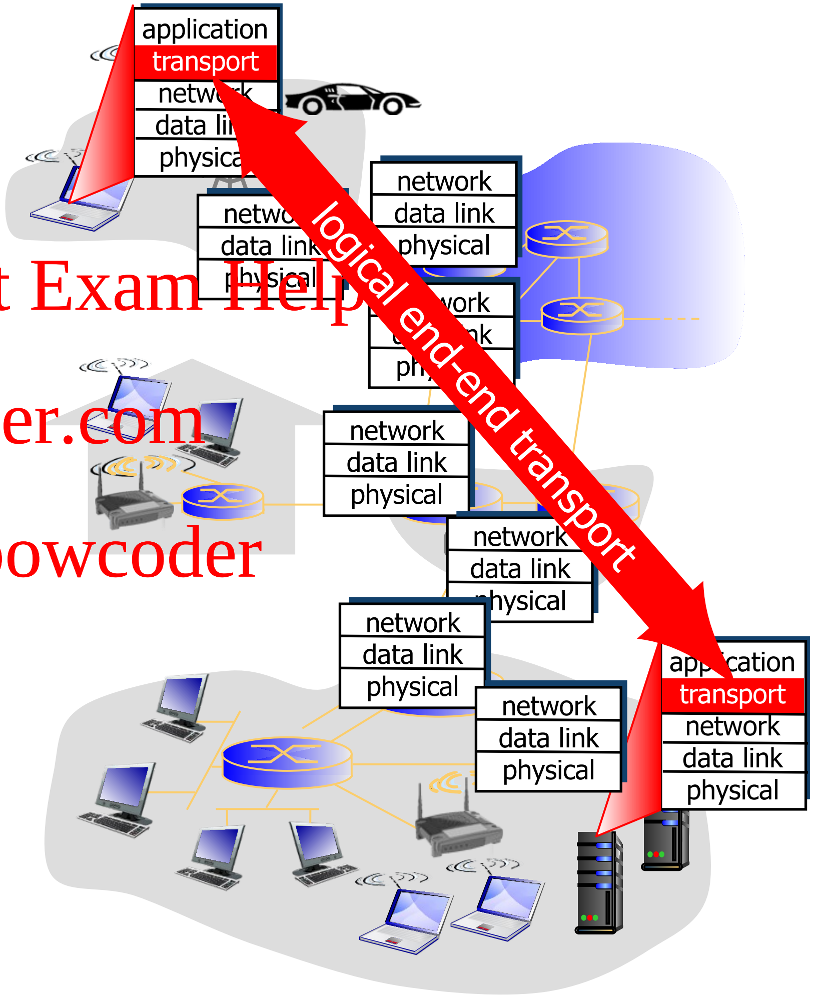
Transport vs. network layer

- *network layer:* host-to-host communication
  - best-effort, unreliable

- *transport layer:* process-to-process communication
  - relies on, enhances, network layer services

Internet transport-layer protocols

- IP: best effort service
- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees

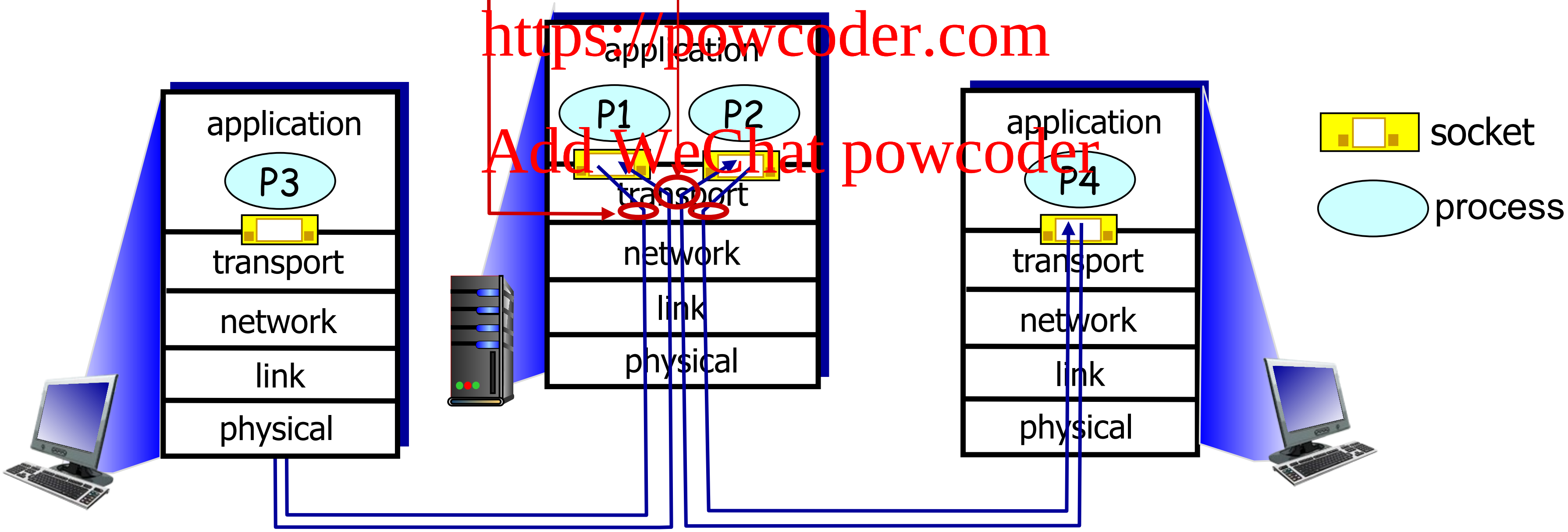


Transport Services

Multiplexing/demultiplexing

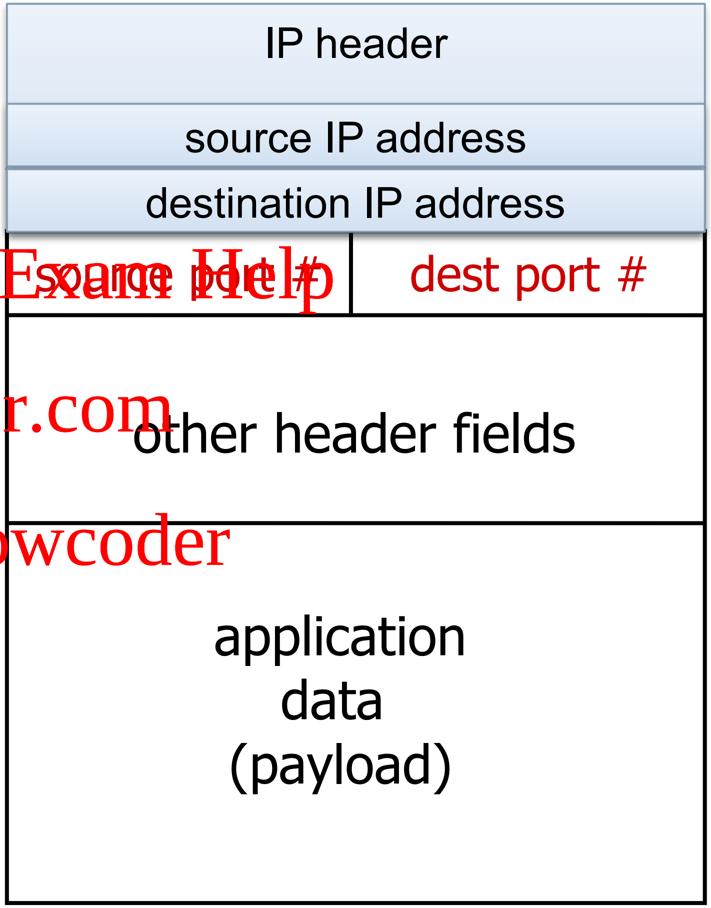
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

demultiplexing at receiver:
use header info to deliver received segments to correct socket



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one transport-layer segment
  - each segment has source, destination port number
- host uses *IP addresses & port numbers* to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

- *Receiver*
- *recall:* created socket has host-local port #:
- when host receives UDP segment:
  - Checks destination port # in segment
  - directs UDP segment to socket with that port #

- *Sender*
- *recall:* when creating datagram to send into UDP socket, must specify
  - destination IP address
  - destination port #

clientSocket.sendto(message,(desip, des port))

Packets with *same dest.IP address, dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at dest

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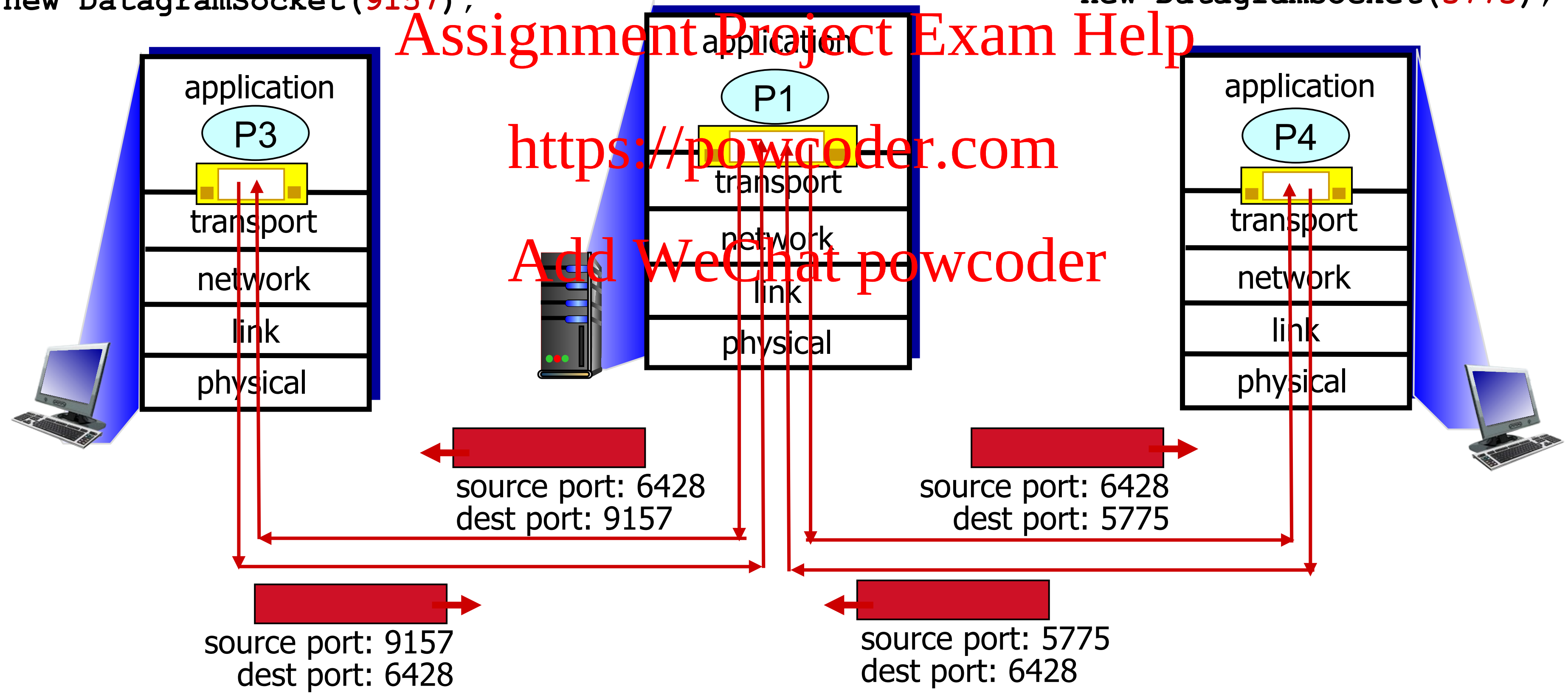
Connectionless demux: example

```
DatagramSocket serverSocket =
 new DatagramSocket(6428);
```

```
DatagramSocket mySocket2 =
new DatagramSocket(9157);
```

```
DatagramSocket mySocket1 =
 new DatagramSocket(5775);
```

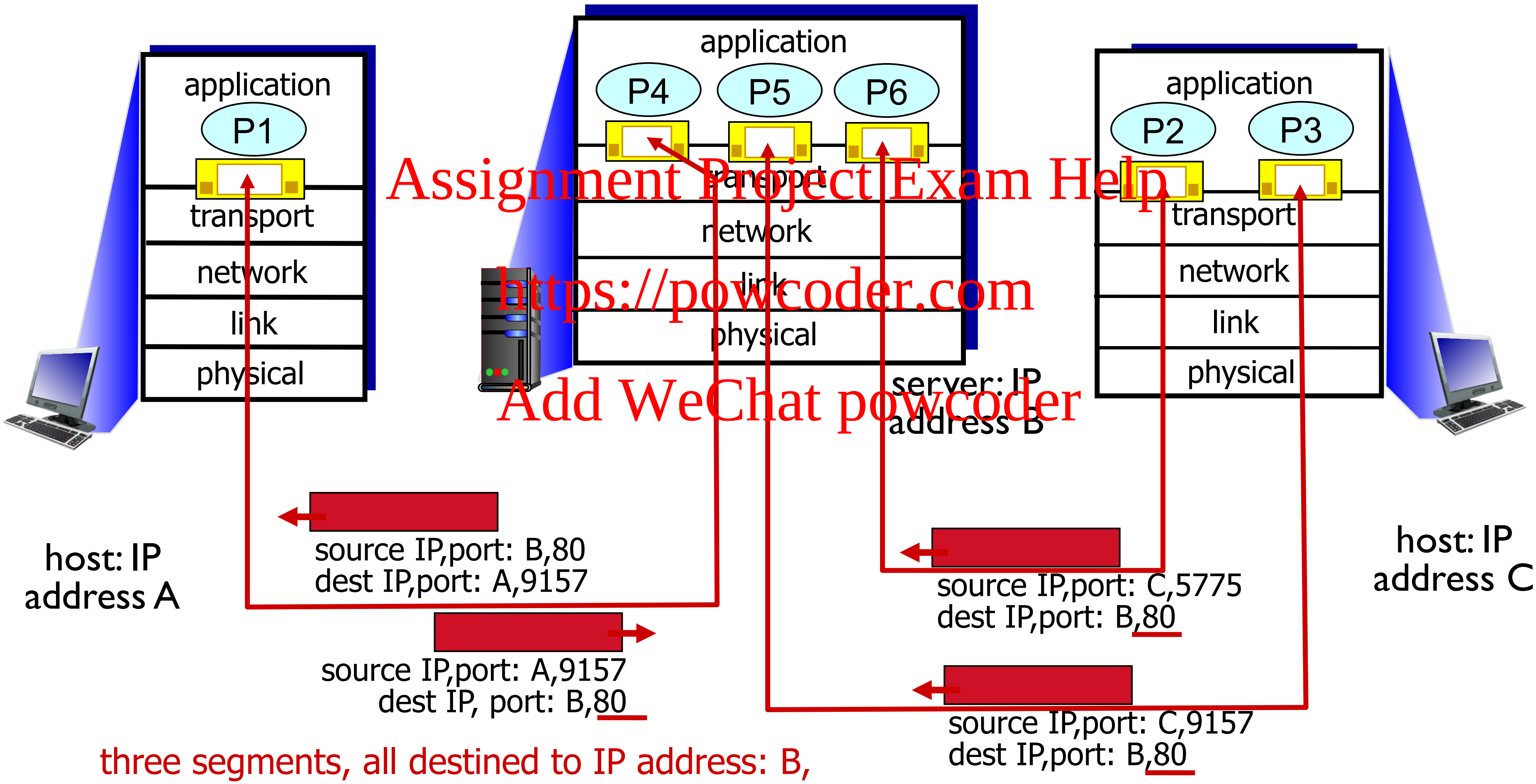


Connection-oriented demux

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket

- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux: example



source IP,port: B,80
dest IP,port: A,9157

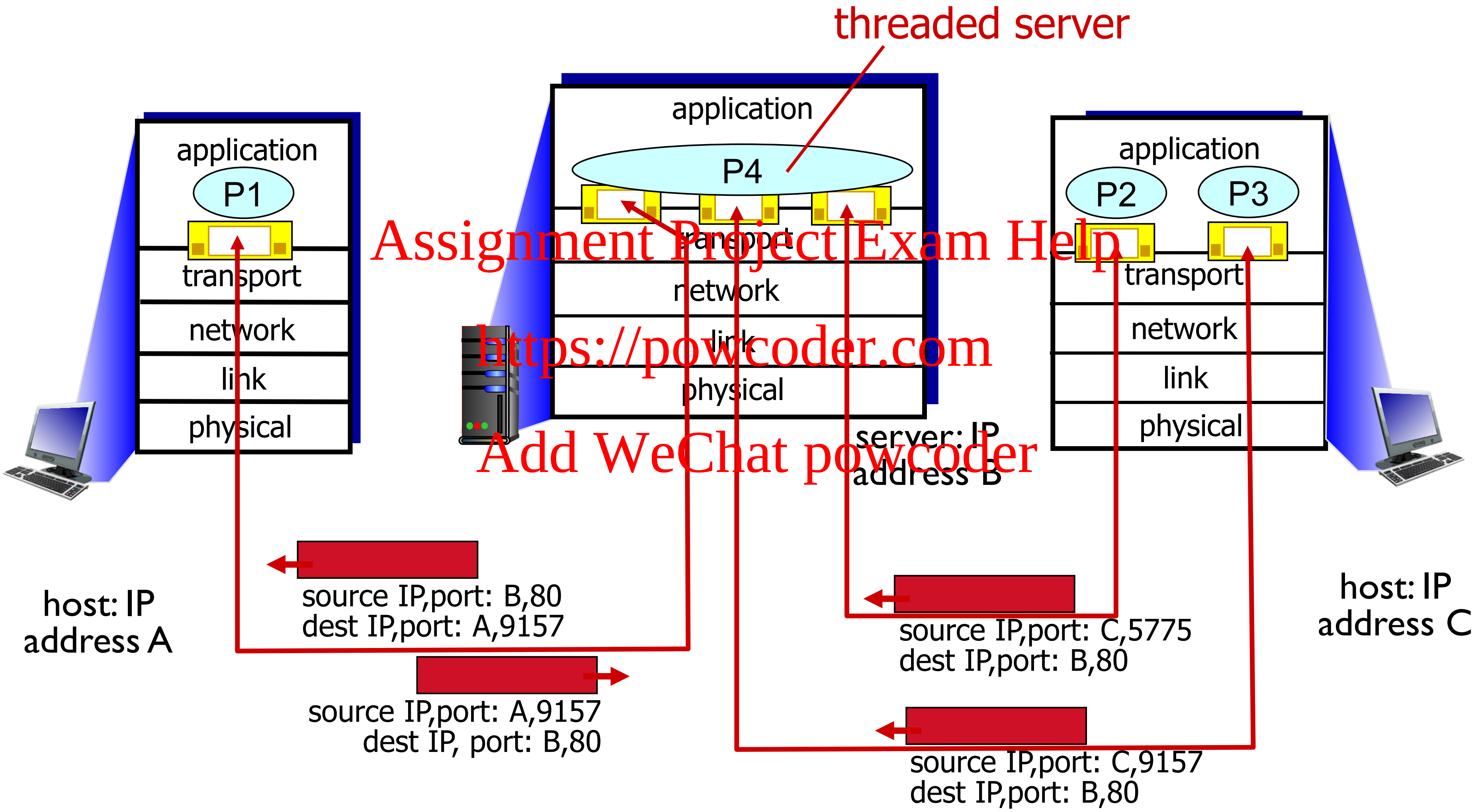
source IP,port: A,9157
dest IP, port: B,80

source IP,port: C,5775
dest IP,port: B,80

source IP,port: C,9157
dest IP,port: B,80

three segments, all destined to IP address: B,
dest port: 80 are demultiplexed to *different* sockets

Connection-oriented demux: example



Connectionless Transport UDP

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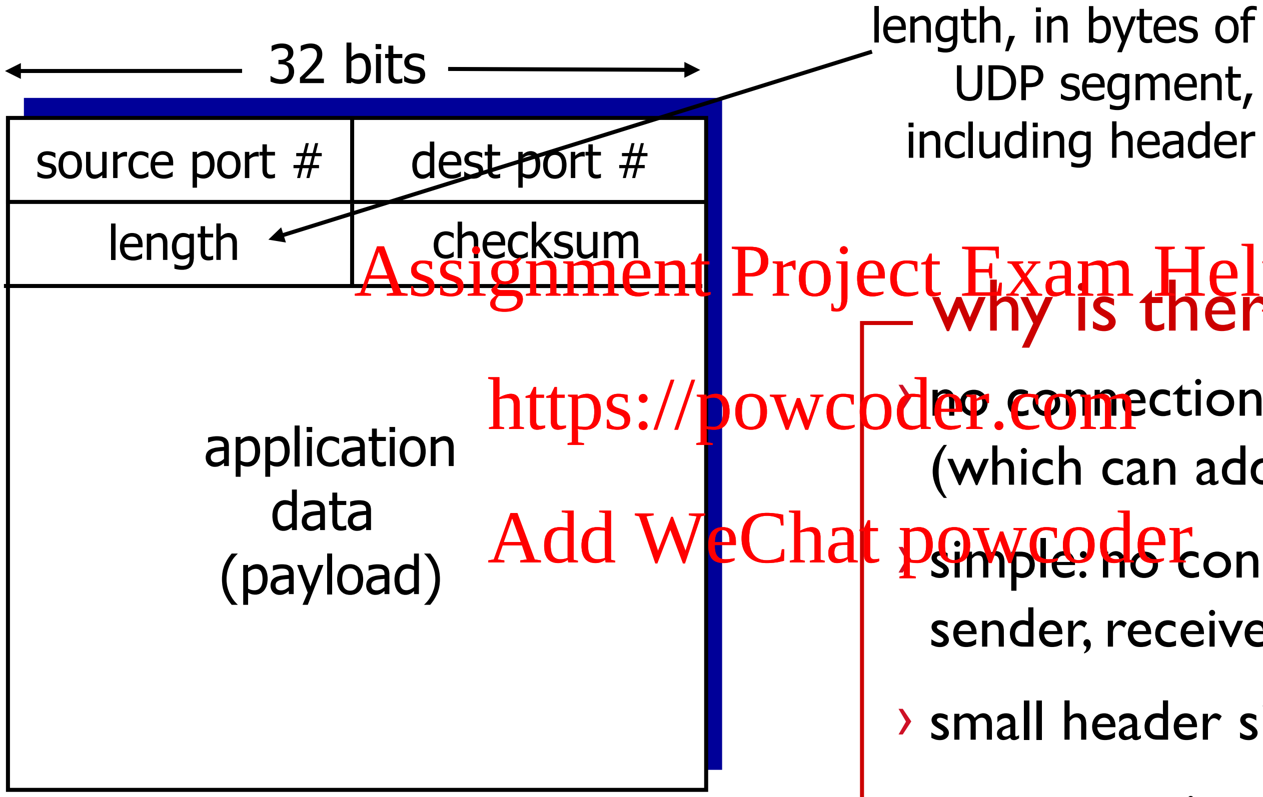
UDP: User Datagram Protocol [RFC 768]

- "no frills," Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!

UDP: segment header


32 bits
source port #
dest port #
length
checksum
application data (payload)
length, in bytes of UDP segment, including header

UDP segment format

why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

UDP checksum

Goal: detect "errors" (e.g., flipped bits) in transmitted segment

sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- sum: addition (one's complement sum) of segment contents
- checksum: complement of sum
- sender puts checksum value into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected.

Internet checksum: example

example: add two 16-bit integers

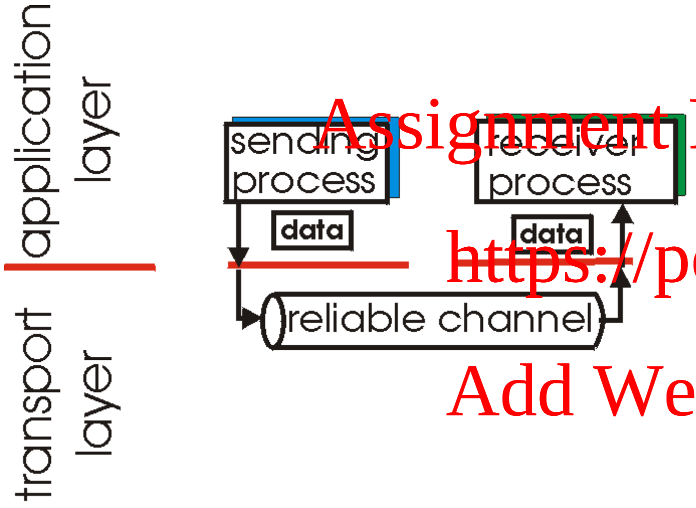
		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
carryout	🟥	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
wraparound		0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

Principles of Reliable Data Transfer

Principles of reliable data transfer

- important in application, transport, link layers
  - top-10 list of important networking topics!

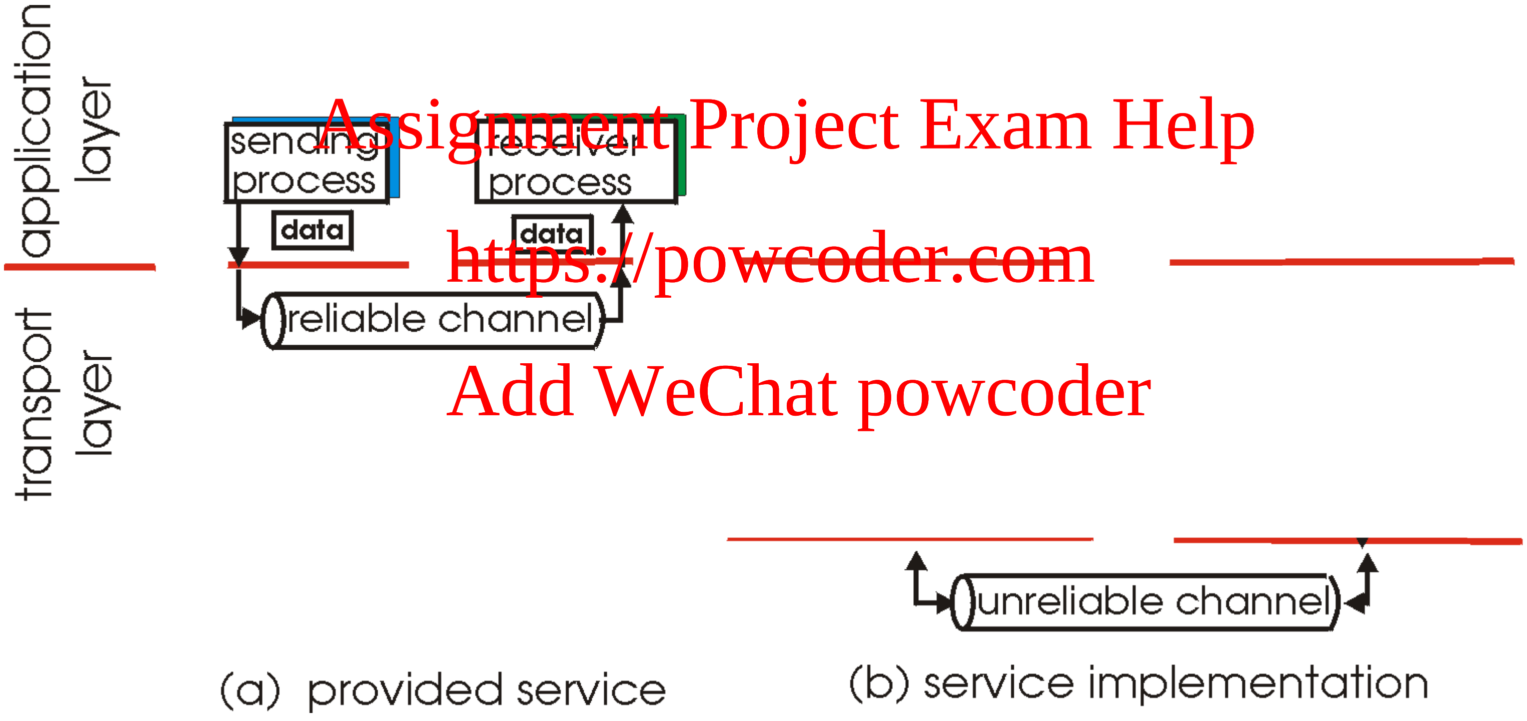


(a) provided service

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of reliable data transfer

- important in application, transport, link layers
  - top-10 list of important networking topics!



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Principles of reliable data transfer

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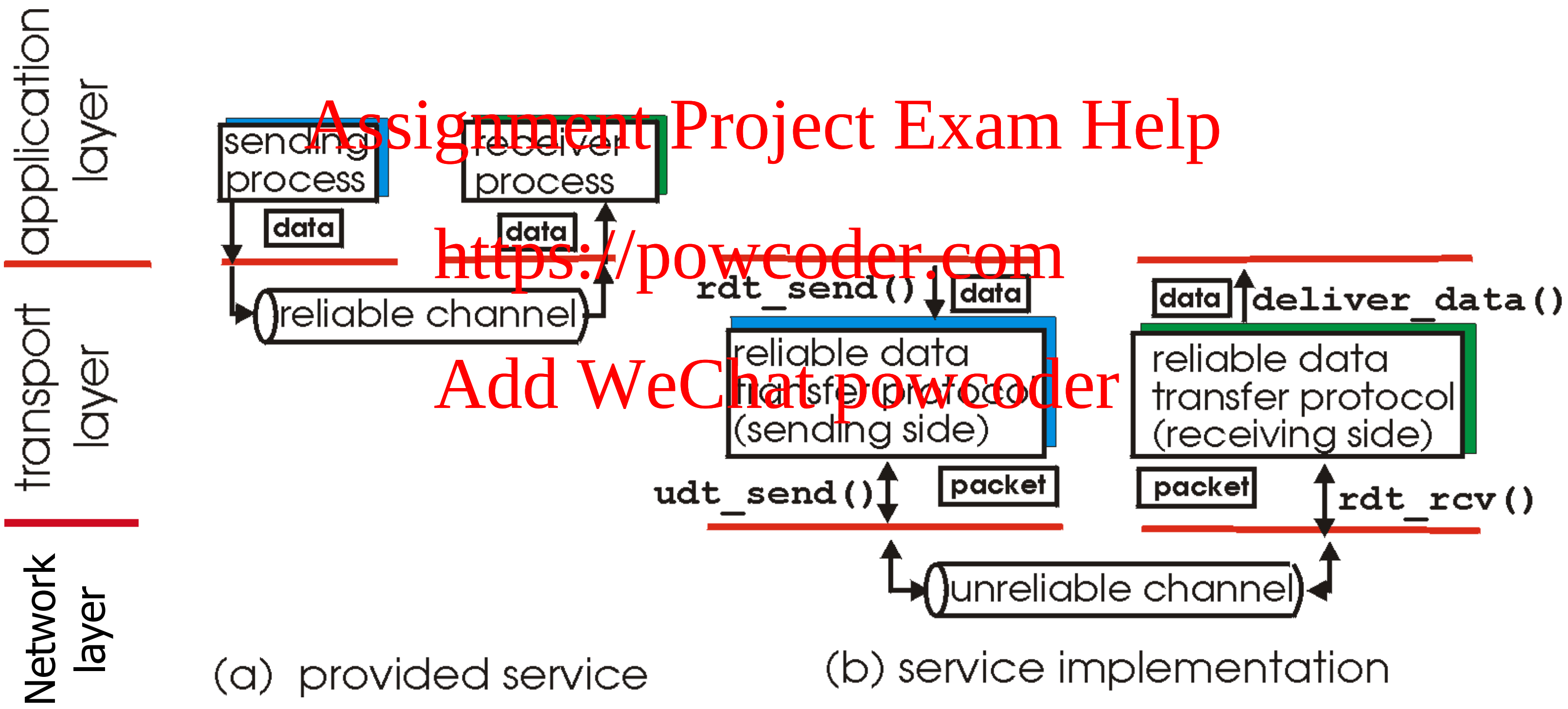
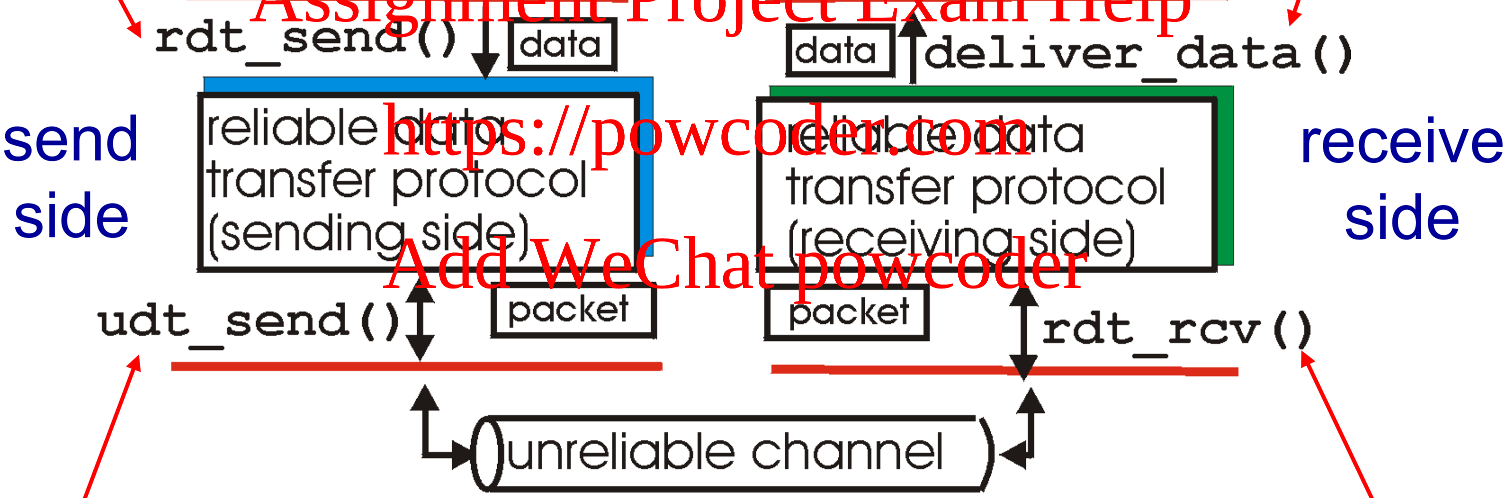


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of reliable data transfer

rdt_send(): called from above, (e.g., by app.). Passed data to deliver to receiver upper layer

deliver_data(): called by **rdt** to deliver data to upper



udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

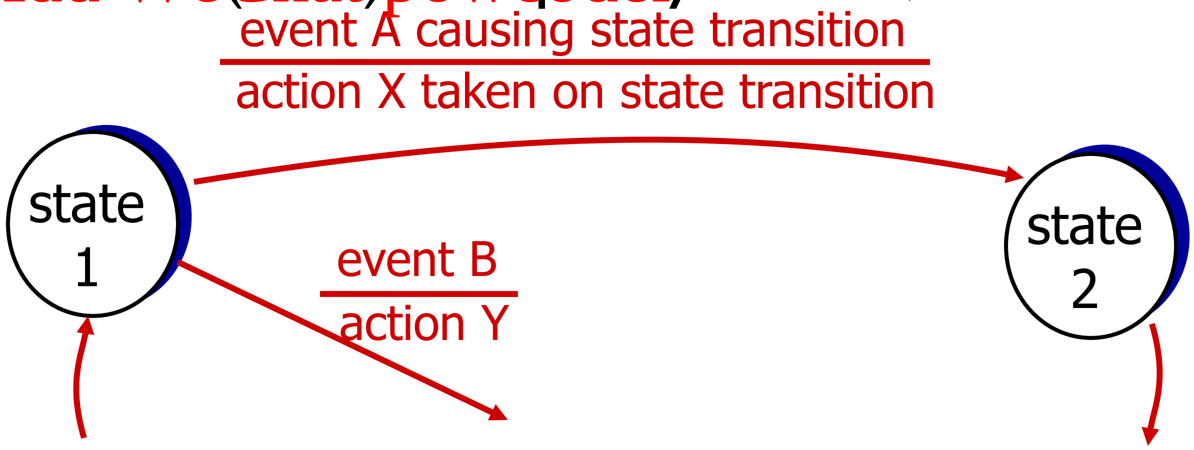
rdt_rcv(): called when packet arrives on rcv-side of channel

Principles of reliable data transfer

We will:

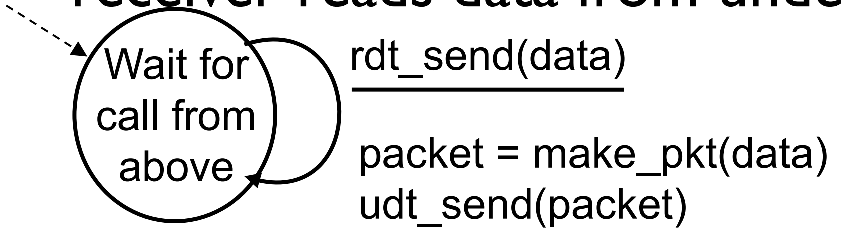
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

state: when in this "state", next state and action uniquely determined by next event

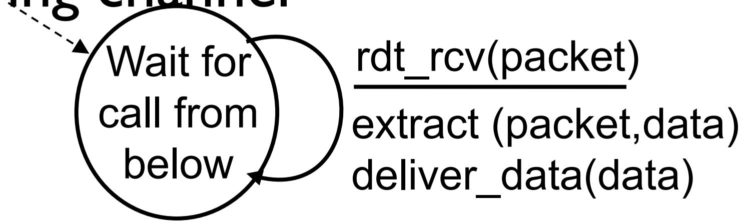

event A causing state transition
action X taken on state transition
state 1
event B
action Y
state 2

Principles of reliable data transfer

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver reads data from underlying channel



sender



receiver

rdt2.0: channel with bit errors

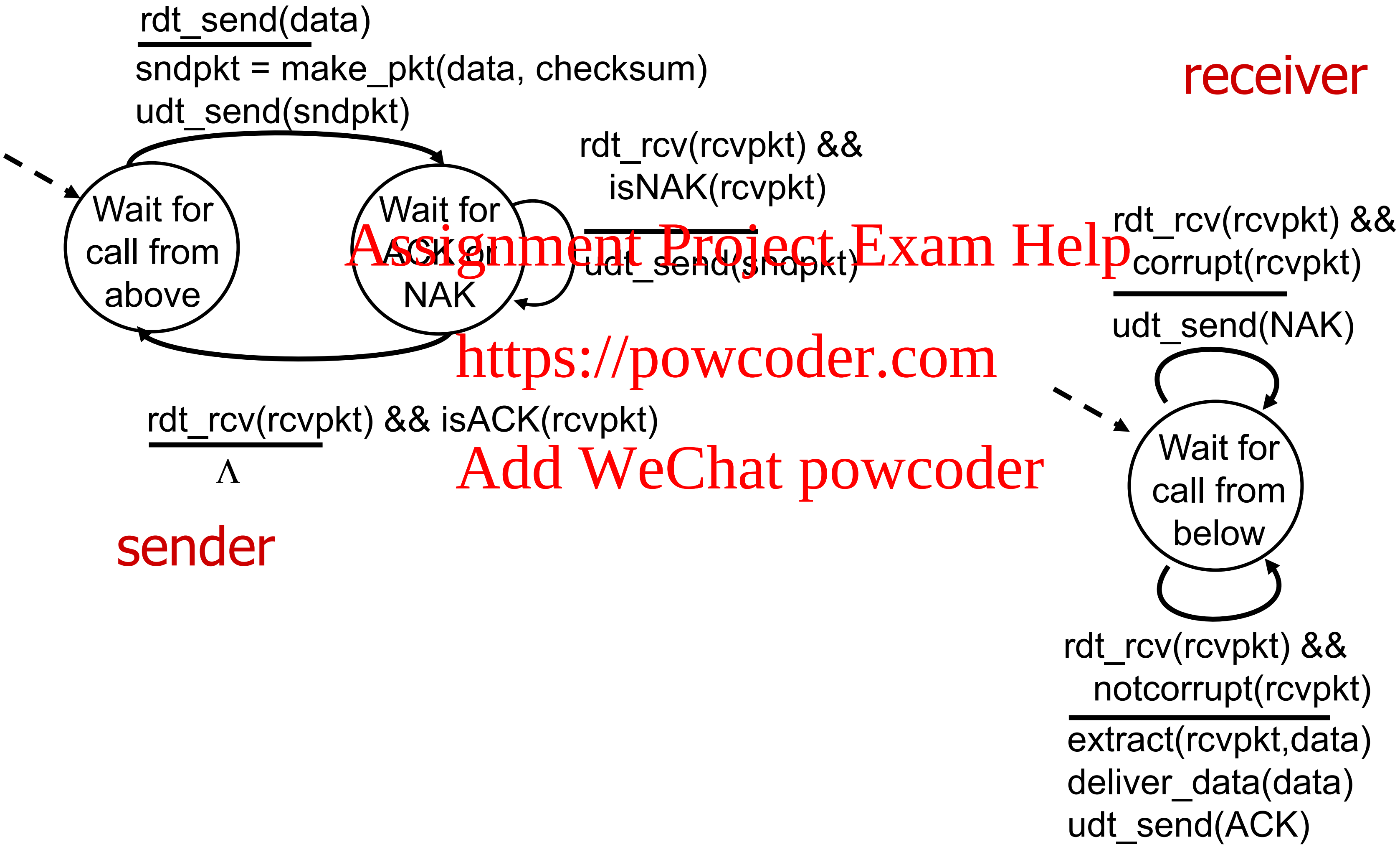
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:

How do humans recover from "errors" during conversation?

rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - sender retransmits pkt on receipt of NAK
- new mechanisms in `rdt2.0` (beyond `rdt1.0`):
  - error detection
  - feedback: control msgs (ACK,NAK) from receiver to sender

rdt2.0: FSM specification



rdt_send(data)

sndpkt = make_pkt(data, checksum)
udt_send(sndpkt)

Wait for call from above

Wait for ACK or NAK

rdt_rcv(rcvpkt) && isNAK(rcvpkt)

udt_send(sndpkt)

rdt_rcv(rcvpkt) && isACK(rcvpkt)

Λ

sender

receiver

rdt_rcv(rcvpkt) && corrupt(rcvpkt)

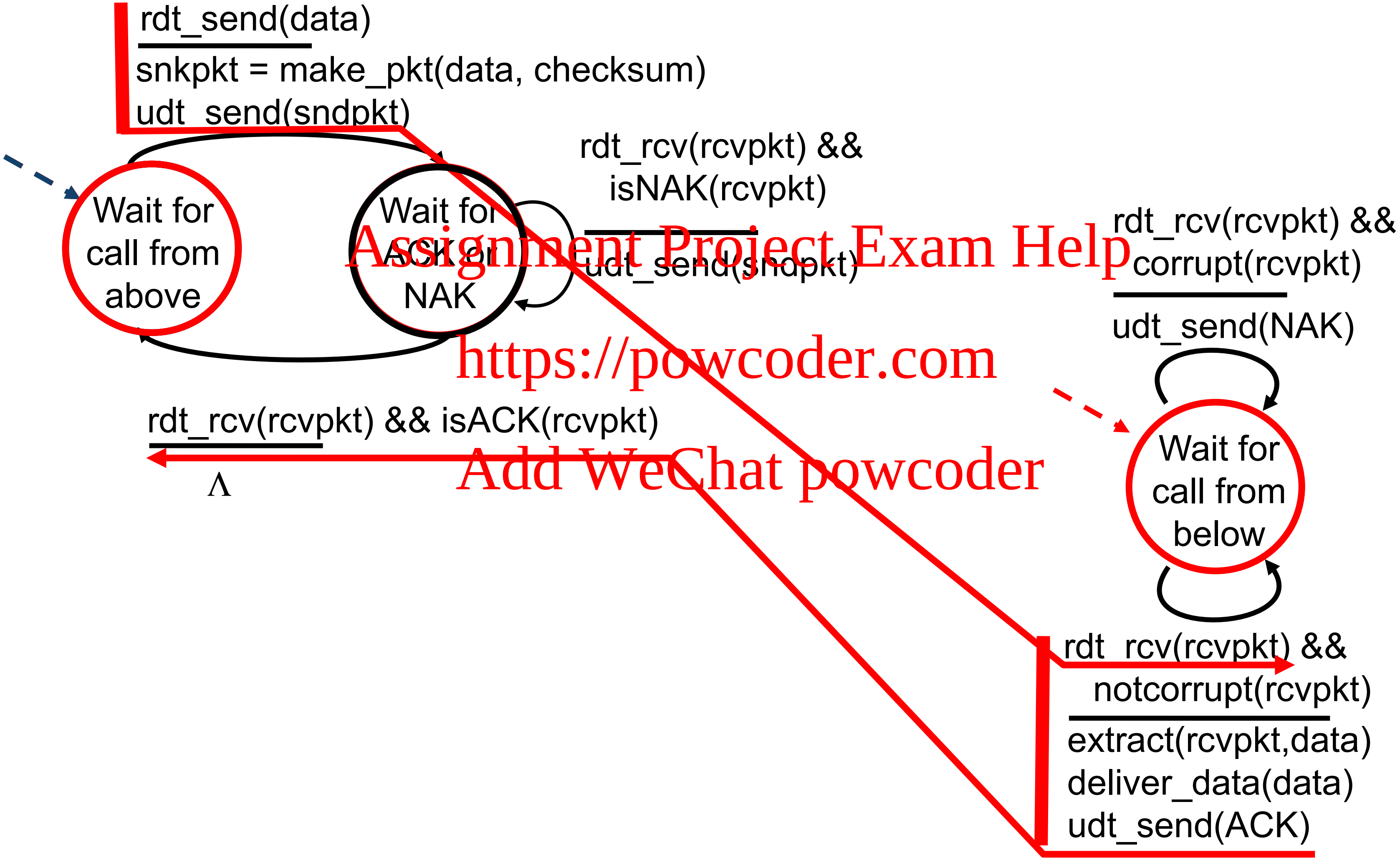
udt_send(NAK)

Wait for call from below

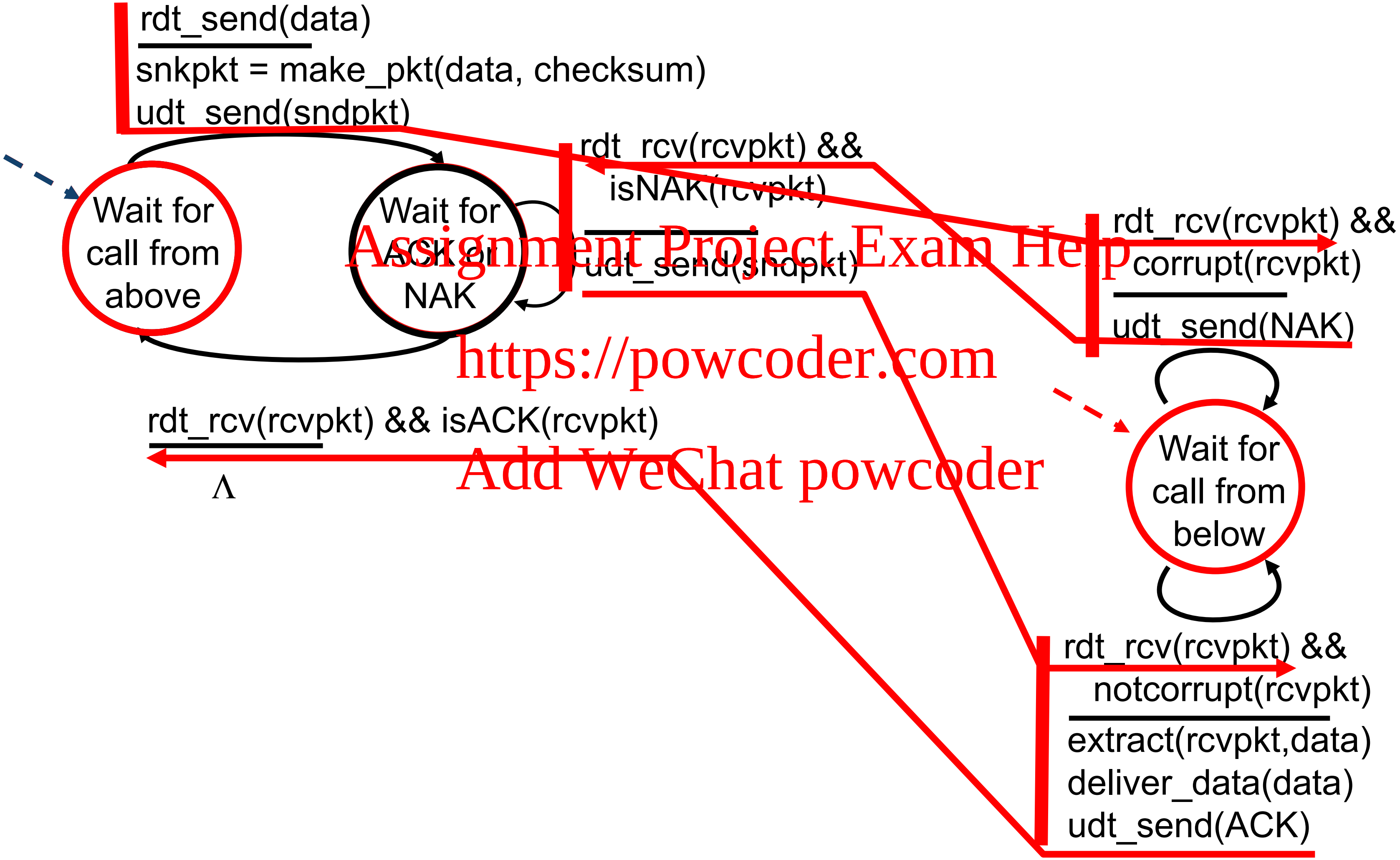
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
udt_send(ACK)

rdt2.0: operation with no errors



rdt2.0: error scenario



rdt_send(data)

snkpkt = make_pkt(data, checksum)
udt_send(sndpkt)

Wait for call from above

Wait for ACK or NAK

rdt_rcv(rcvpkt) && isNAK(rcvpkt)

udt_send(sndpkt)

rdt_rcv(rcvpkt) && isACK(rcvpkt)

Λ

rdt_rcv(rcvpkt) && corrupt(rcvpkt)

udt_send(NAK)

Wait for call from below

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
udt_send(ACK)

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rdt2.0 has a fatal flaw!

what happens if ACK/NAK corrupted?

- sender does not know what happened at receiver!
- cannot just retransmit: possible duplicate

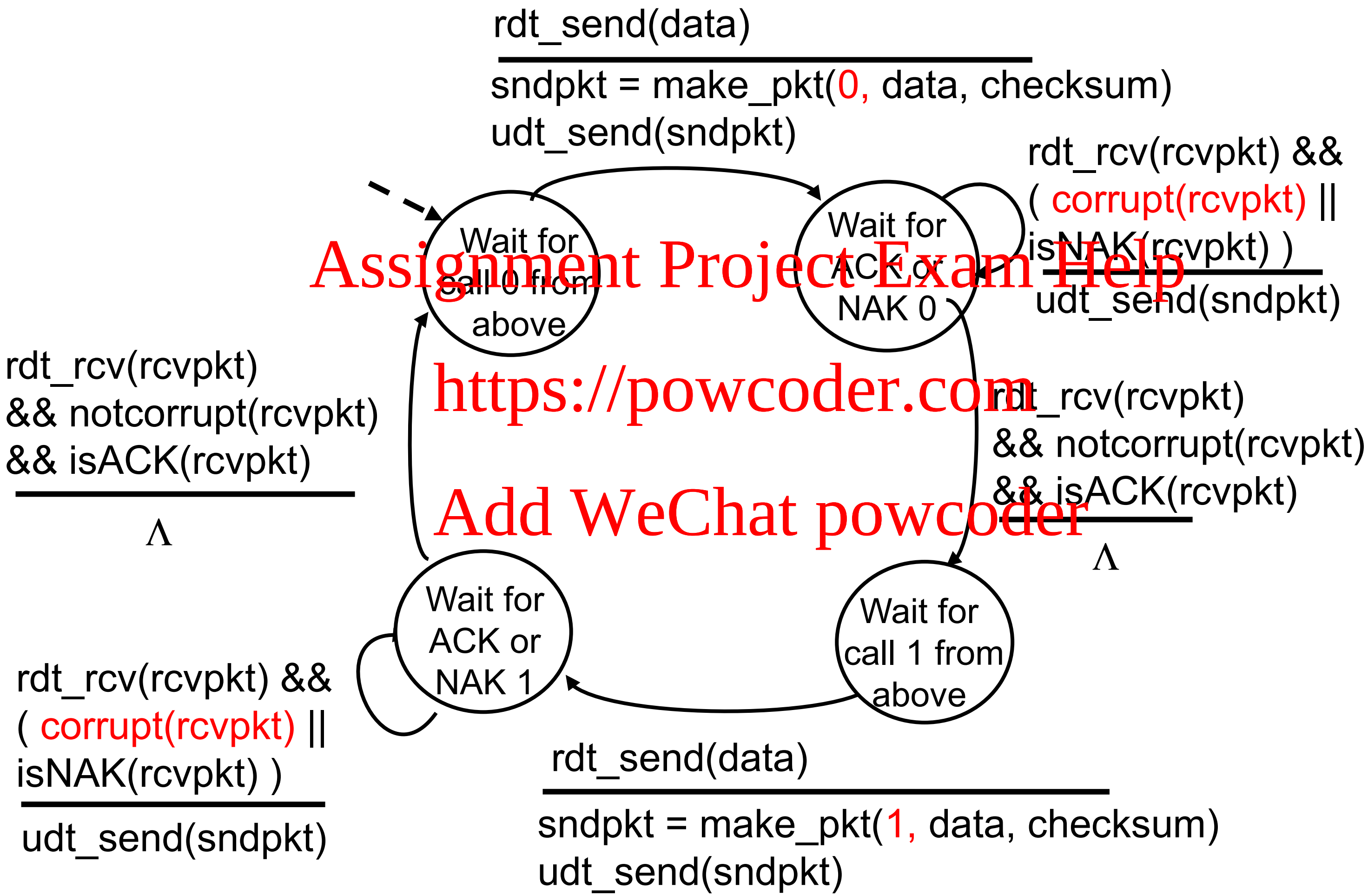
handling duplicates:

- sender retransmits current pkt if ACK/NAK corrupted
- sender adds *sequence number* to each pkt
- receiver discards (does not deliver up) duplicate pkt

stop and wait

sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

Wait for call 0 from above

Wait for ACK or NAK 0

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isNAK(rcvpkt))

udt_send(sndpkt)

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt)

Λ

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt)

Λ

Wait for ACK or NAK 1

Wait for call 1 from above

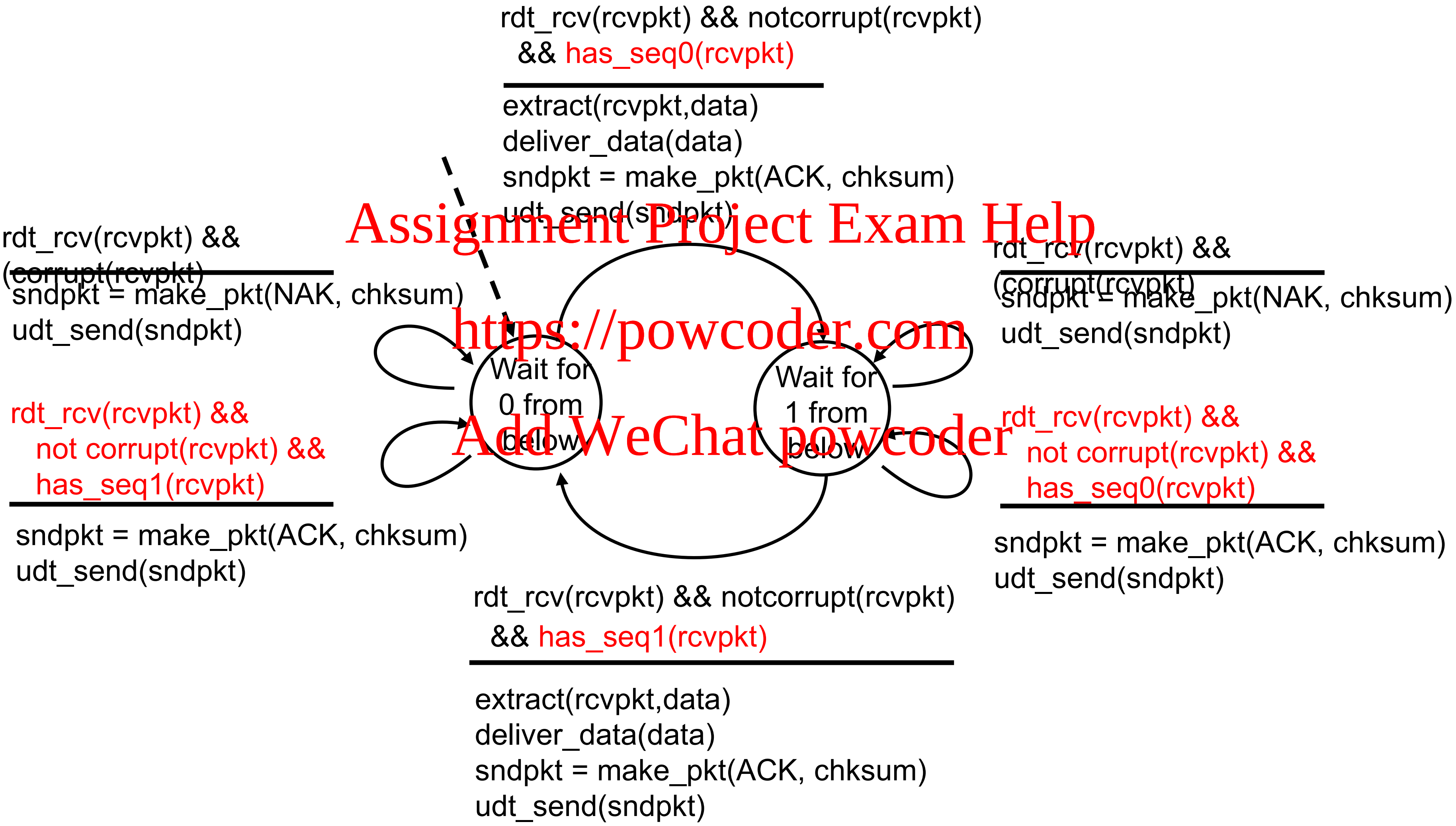
rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isNAK(rcvpkt))

udt_send(sndpkt)

rdt_send(data)

sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)

rdt2.1: receiver, handles garbled ACK/NAKs



rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq0(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && (corrupt(rcvpkt)

sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
not corrupt(rcvpkt) &&
has_seq1(rcvpkt)

sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && (corrupt(rcvpkt)

sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
not corrupt(rcvpkt) &&
has_seq0(rcvpkt)

sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt2.1: discussion

<u>sender:</u>

- seq # added to pkt
- two seq. #'s (0,1) will suffice.
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "expected" pkt should have seq # of 0 or 1

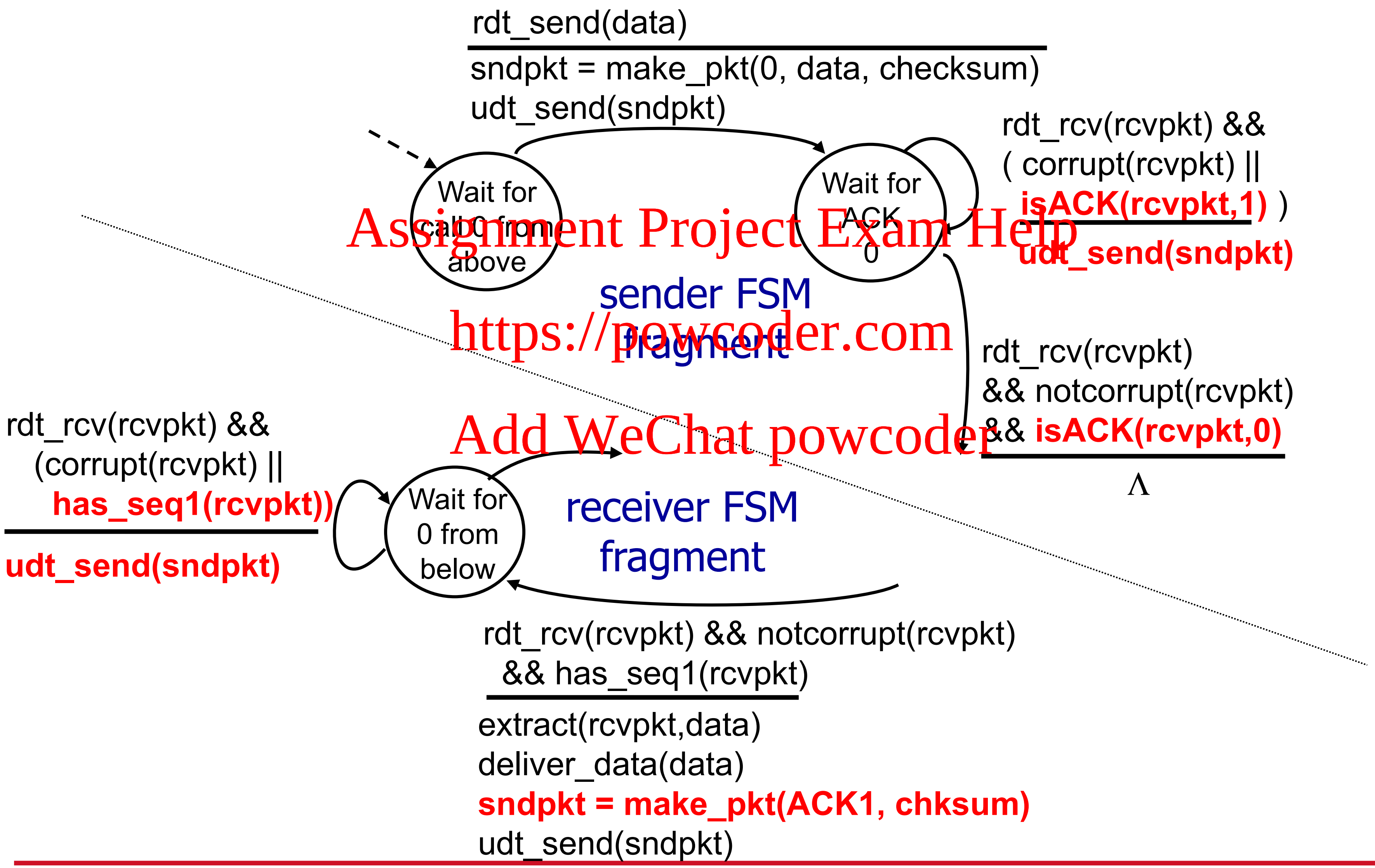
<u>receiver:</u>

- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #

rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last pkt received OK
  - receiver must *explicitly* include seq # of pkt being ACKed
- "unexpected" ACK at sender results in same action as NAK: *retransmit current pkt*

rdt2.2: sender, receiver fragments



rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

Wait for call 0 from above

Wait for ACK 0

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))

udt_send(sndpkt)

sender FSM fragment

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& **isACK(rcvpkt,0)**

Λ

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
has_seq1(rcvpkt))

udt_send(sndpkt)

Wait for 0 from below

receiver FSM fragment

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK1, chksum)
udt_send(sndpkt)

rdt3.0: channels with errors and loss

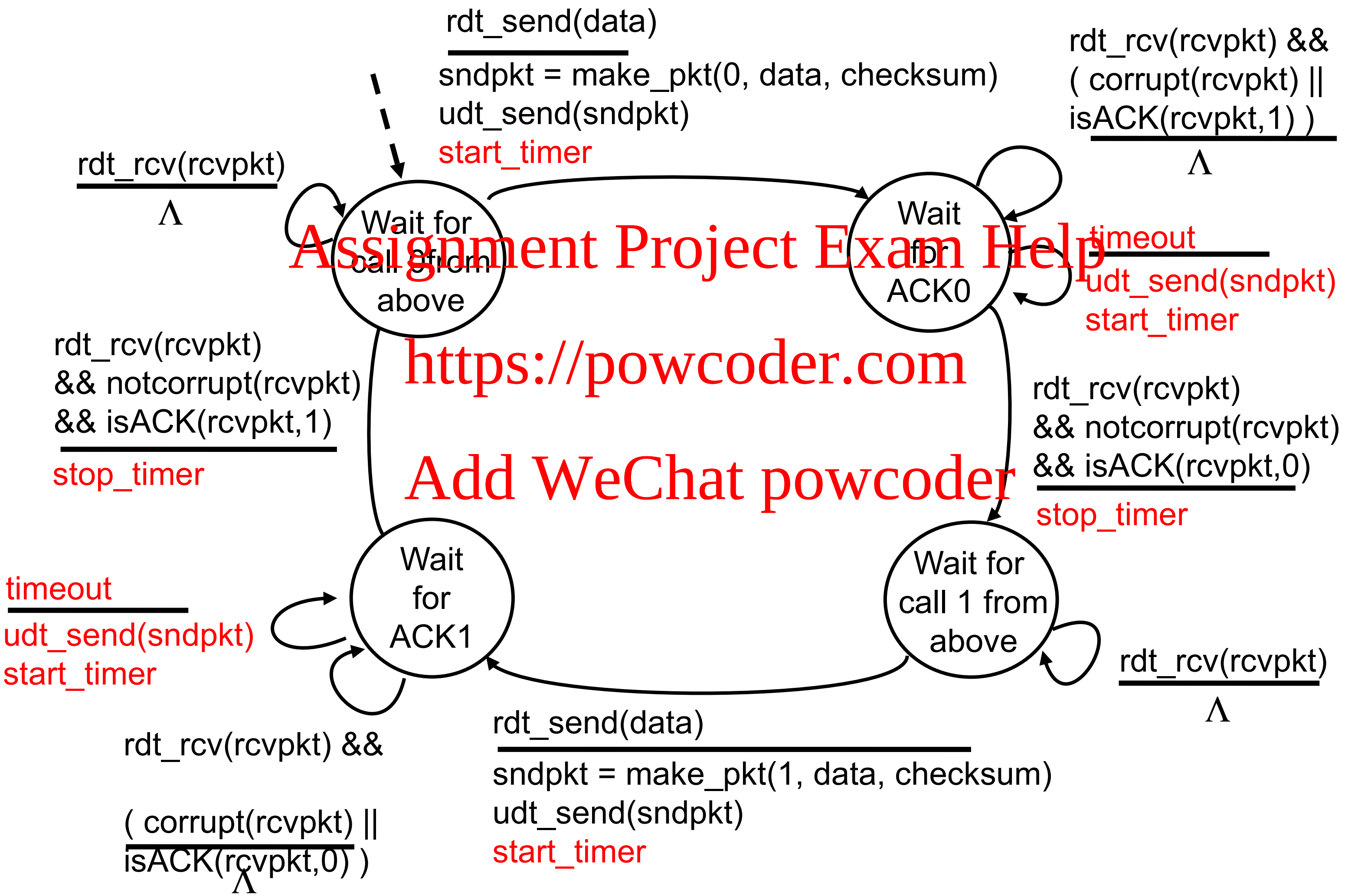
new assumption: underlying channel can also lose packets (data, ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help … but not enough

approach: sender waits "reasonable" amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

rdt3.0 sender



rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))

Λ

rdt_rcv(rcvpkt)

Λ

Wait for call 0 from above

Wait for ACK0

timeout

udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,1)

stop_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,0)

stop_timer

timeout

udt_send(sndpkt)
start_timer

Wait for ACK1

Wait for call 1 from above

rdt_rcv(rcvpkt)

Λ

rdt_rcv(rcvpkt) &&

(corrupt(rcvpkt) ||
isACK(rcvpkt,0))

Λ

rdt_send(data)

sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)
start_timer

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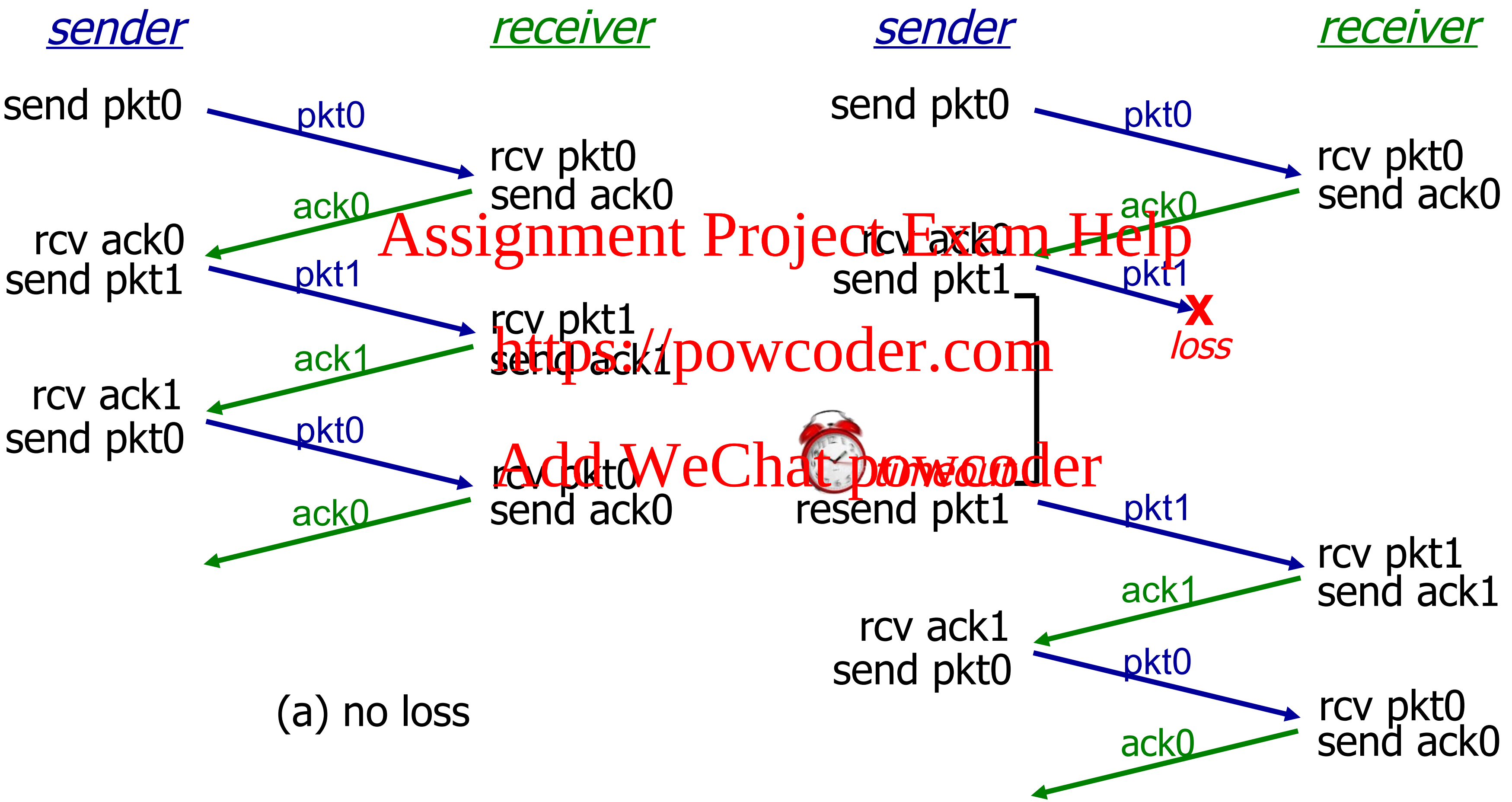
rdt3.0 in action



sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

rcv pkt1
send ack1

ack1

rcv ack1
send pkt0

pkt0

rcv pkt0
send ack0

ack0

(a) no loss

sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

X
loss

timeout
resend pkt1

pkt1

rcv pkt1
send ack1

ack1

rcv ack1
send pkt0

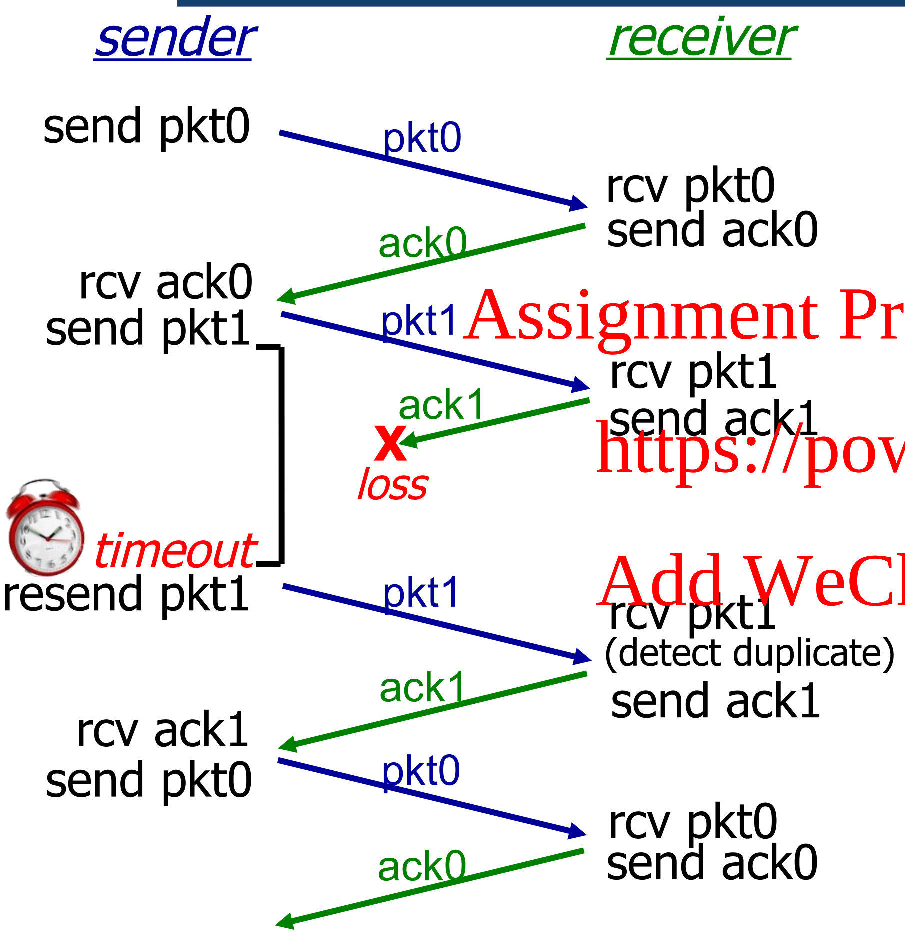
pkt0

rcv pkt0
send ack0

ack0

(b) packet loss

rdt3.0 in action


sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

rcv pkt1
send ack1

ack1

X
loss

timeout

resend pkt1

pkt1

rcv pkt1
(detect duplicate)
send ack1

ack1

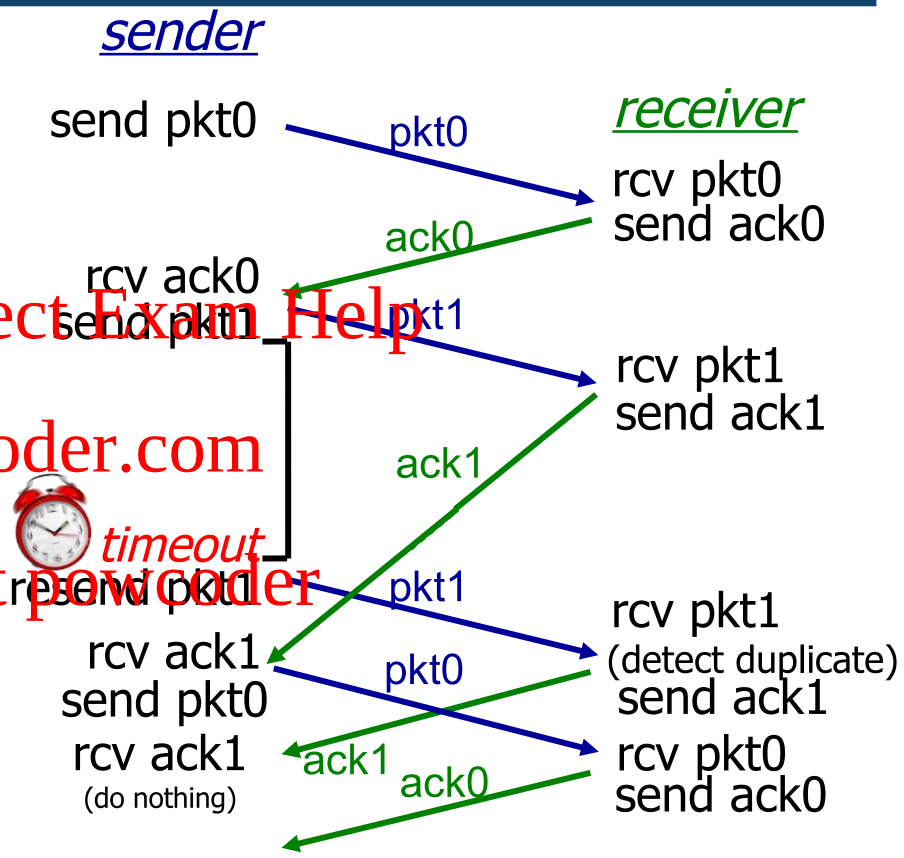
rcv ack1
send pkt0

pkt0

rcv pkt0
send ack0

ack0

(c) ACK loss


sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

rcv pkt1
send ack1

ack1

timeout

resend pkt1

pkt1

rcv pkt1
(detect duplicate)
send ack1

rcv ack1
send pkt0

pkt0

rcv pkt0
send ack0

ack1

rcv ack1
(do nothing)

ack0

(d) premature timeout/ delayed ACK

Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

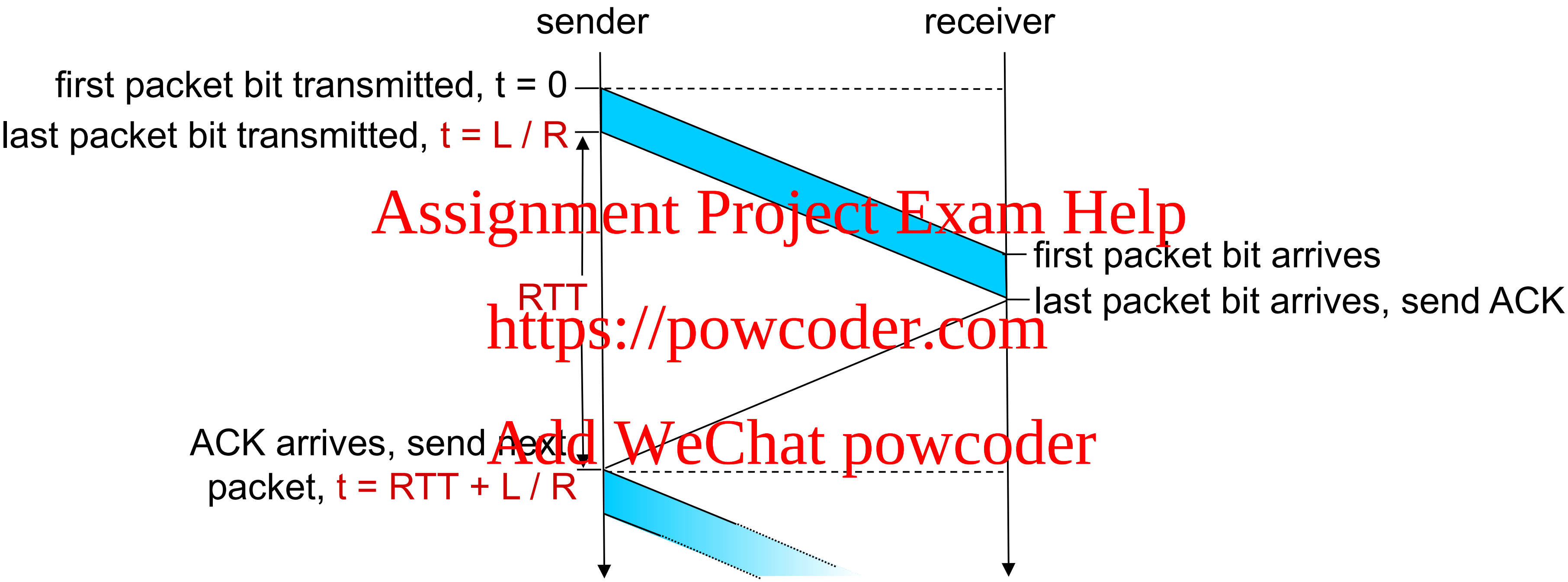
- U_{sender}: *utilization* – fraction of time sender busy sending

$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

- if RTT=30 msec, 1KB pkt every 30 msec: 33kB/sec thruput over 1 Gbps link

- network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation



sender

receiver

first packet bit transmitted, t = 0

last packet bit transmitted, t = L / R

RTT

first packet bit arrives

last packet bit arrives, send ACK

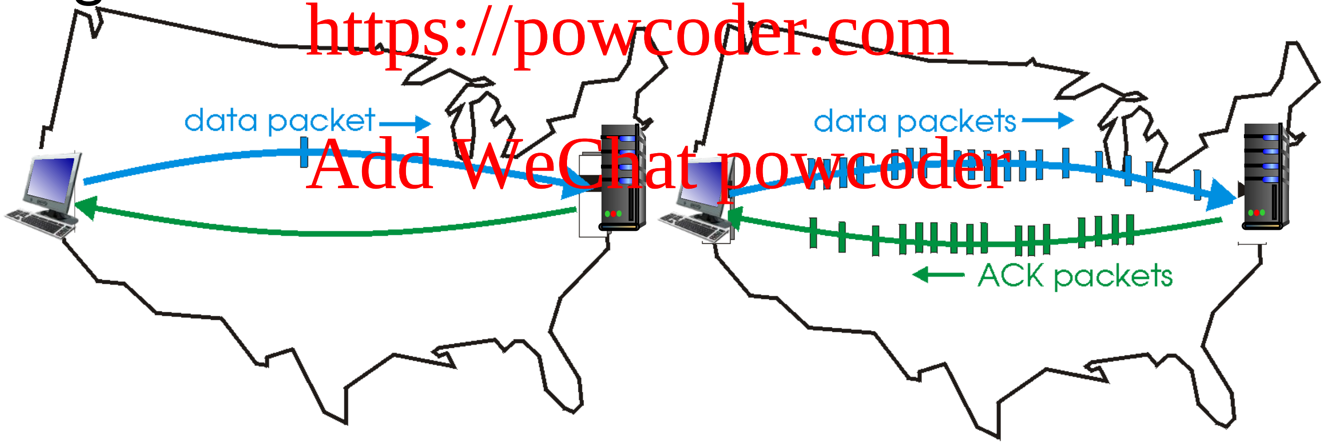
ACK arrives, send next packet, t = RTT + L / R

$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver

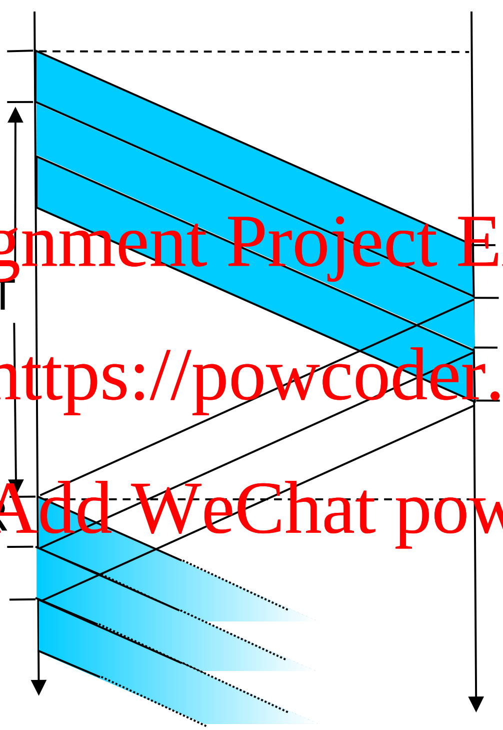


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

› two generic forms of pipelined protocols: *go-Back-N, selective repeat*

Pipelining: increased utilization



sender
receiver
first packet bit transmitted, t = 0
last bit transmitted, t = L / R
RTT
first packet bit arrives
last packet bit arrives, send ACK
last bit of 2nd packet arrives, send ACK
last bit of 3rd packet arrives, send ACK
ACK arrives, send next packet, t = RTT + L / R

3-packet pipelining increases utilization by a factor of 3!

$$U_{\text{sender}} = \frac{3L / R}{RTT + L / R} = \frac{.0024}{30.008} = 0.00081$$

Pipelined protocols: overview

Go-back-N:

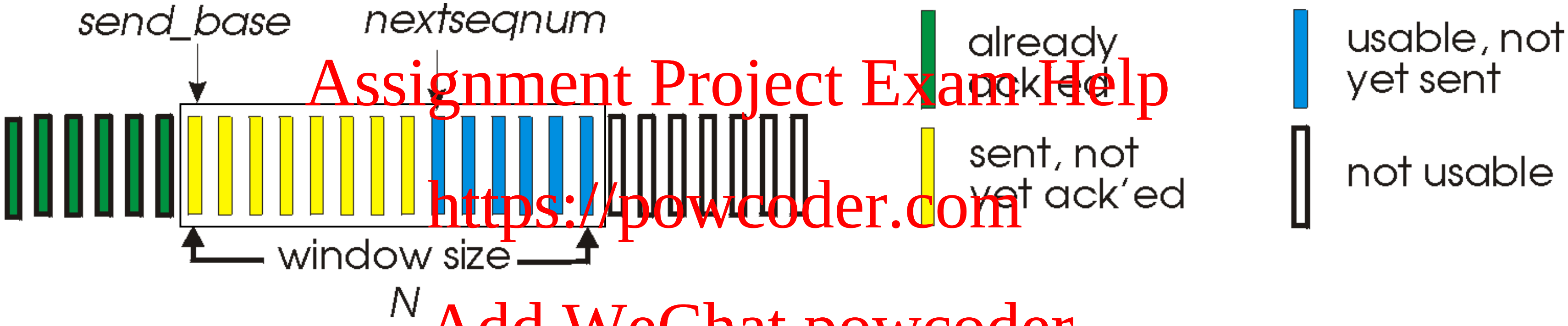
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - does not ack packet if there is a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N unacked packets in pipeline
- receiver sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only that unacked packet

Go-Back-N: sender

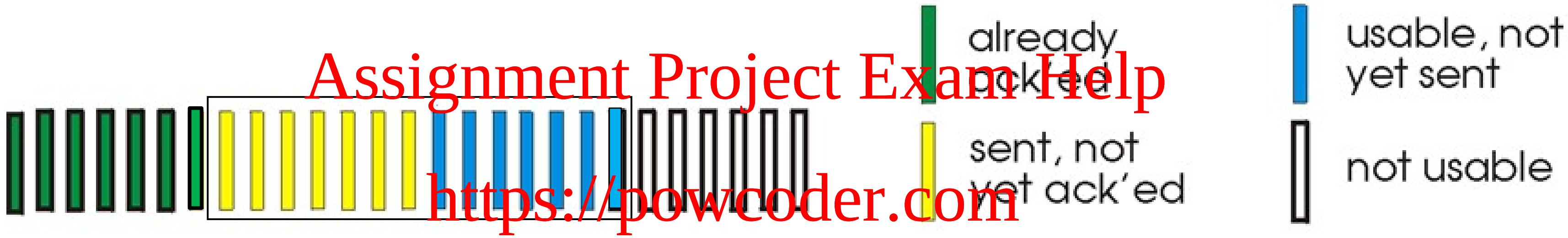
› "window" of up to N, consecutive unacked pkts allowed



- ACK(n): ACKs all pkts up to, including seq # n - *"cumulative ACK"*
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n)*: retransmit packet n and all higher seq # pkts in window

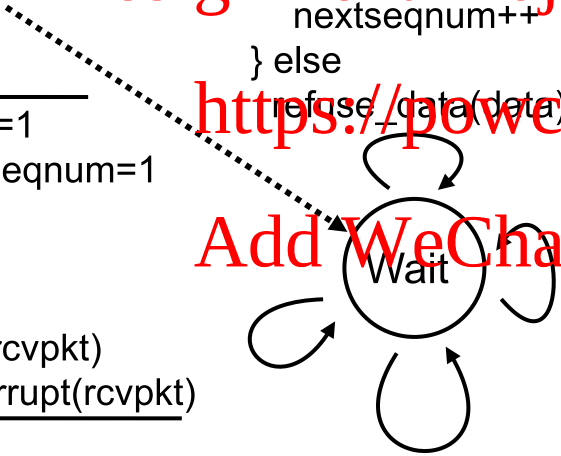
Go-Back-N: sender

- “window” of up to N, consecutive unacked pkts allowed



- ACK(n): ACKs all pkts up to, including seq # n - *“cumulative ACK”*
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n):* retransmit packet n and all higher seq # pkts in window

GBN: sender extended FSM



```
rdt_send(data)

if (nextseqnum < base+N) {
    sndpkt[nextseqnum] = make_pkt(nextseqnum,data,chksum)
    udt_send(sndpkt[nextseqnum])
    if (base == nextseqnum)
       start_timer
    nextseqnum++
    }
else
  refuse_data(data)
```

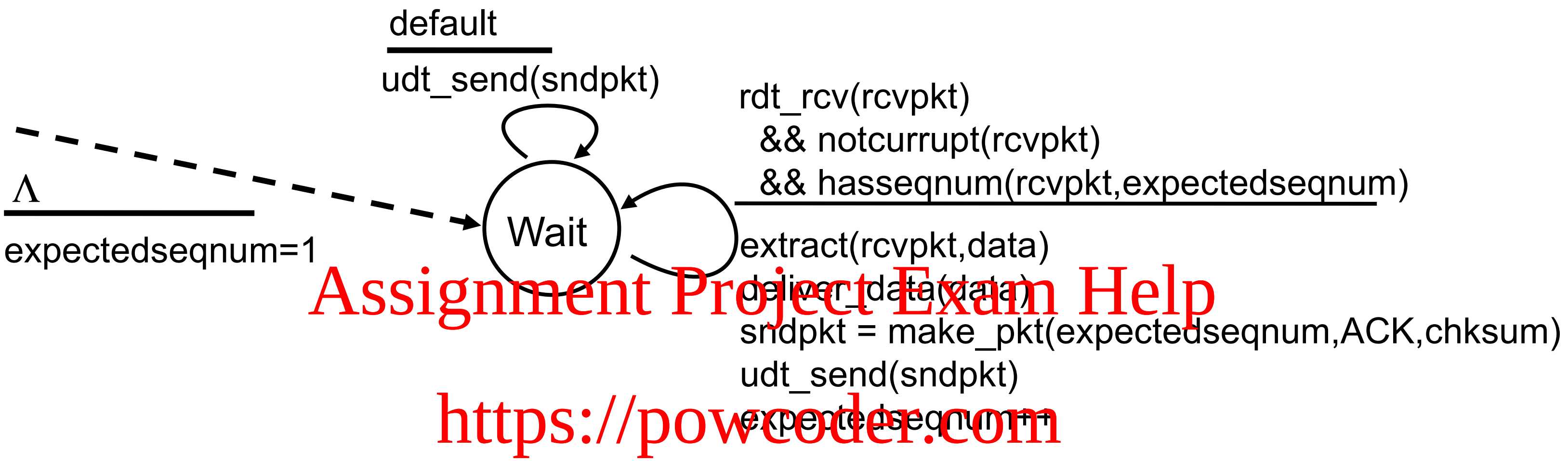
```
Λ

base=1
nextseqnum=1
```

```
timeout

start_timer
udt_send(sndpkt[base])
udt_send(sndpkt[base+1])
…
udt_send(sndpkt[nextseqnum-1])
```

```
rdt_rcv(rcvpkt)
   && corrupt(rcvpkt)
```

```
rdt_rcv(rcvpkt) &&
   notcorrupt(rcvpkt)

base = getacknum(rcvpkt)+1
If (base == nextseqnum)
    stop_timer
else
    start_timer
```

GBN: receiver extended FSM



default
udt_send(sndpkt)

Λ
expectedseqnum=1

Wait

rdt_rcv(rcvpkt)
  && notcurrupt(rcvpkt)
  && hasseqnum(rcvpkt,expectedseqnum)

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(expectedseqnum,ACK,chksum)
udt_send(sndpkt)
expectedseqnum++

ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- may generate duplicate ACKs
- need only remember **expectedseqnum**

› out-of-order pkt:

- discard (don't buffer): *no receiver buffering!*
- re-ACK pkt with highest in-order seq #

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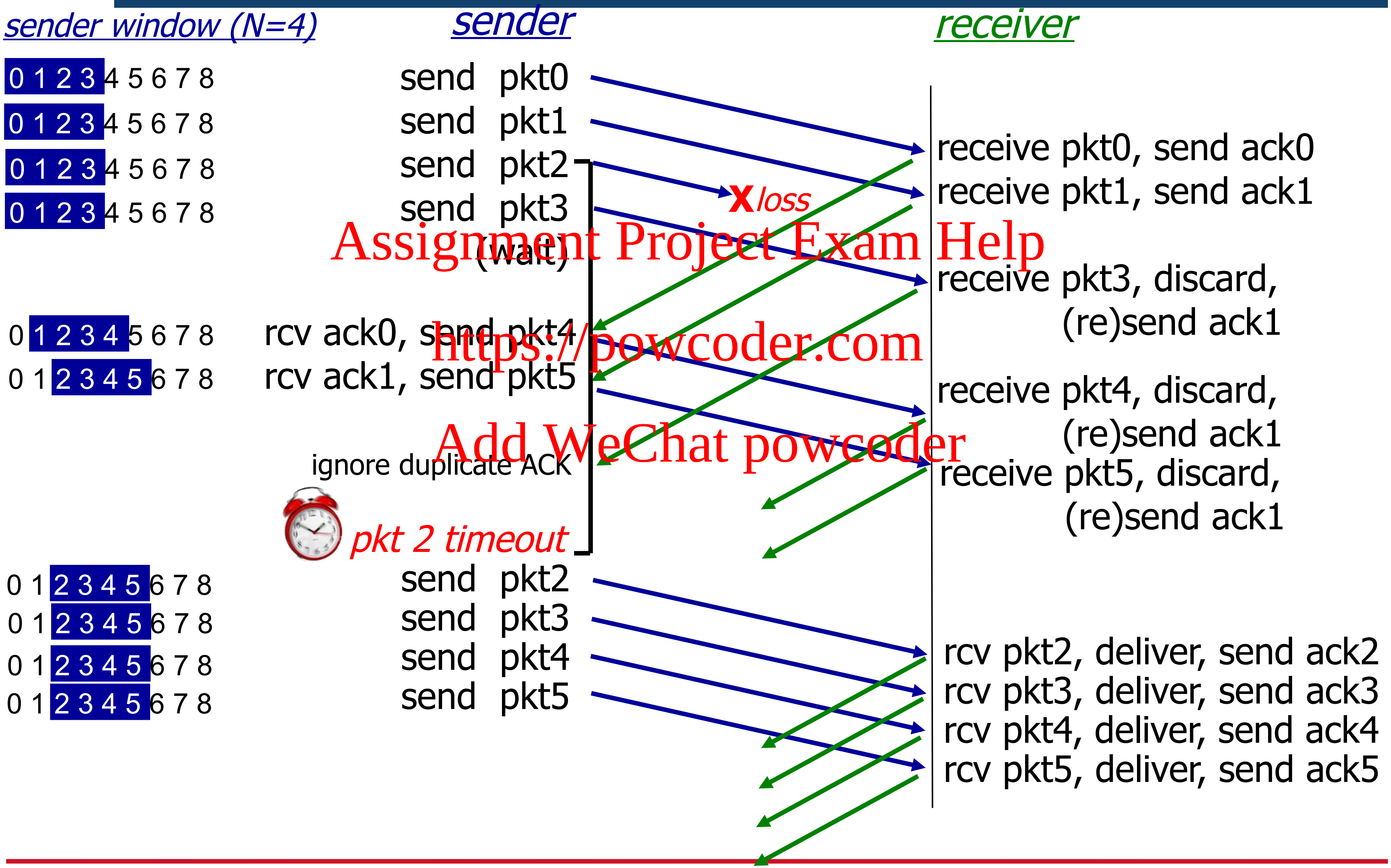
GBN in action



sender window (N=4)

sender

receiver

0 1 2 3 4 5 6 7 8 send pkt0

0 1 2 3 4 5 6 7 8 send pkt1

0 1 2 3 4 5 6 7 8 send pkt2

0 1 2 3 4 5 6 7 8 send pkt3

(wait)

X loss

receive pkt0, send ack0

receive pkt1, send ack1

receive pkt3, discard, (re)send ack1

0 1 2 3 4 5 6 7 8 rcv ack0, send pkt4

0 1 2 3 4 5 6 7 8 rcv ack1, send pkt5

receive pkt4, discard, (re)send ack1

receive pkt5, discard, (re)send ack1

ignore duplicate ACK

pkt 2 timeout

0 1 2 3 4 5 6 7 8 send pkt2

0 1 2 3 4 5 6 7 8 send pkt3

0 1 2 3 4 5 6 7 8 send pkt4

0 1 2 3 4 5 6 7 8 send pkt5

rcv pkt2, deliver, send ack2

rcv pkt3, deliver, send ack3

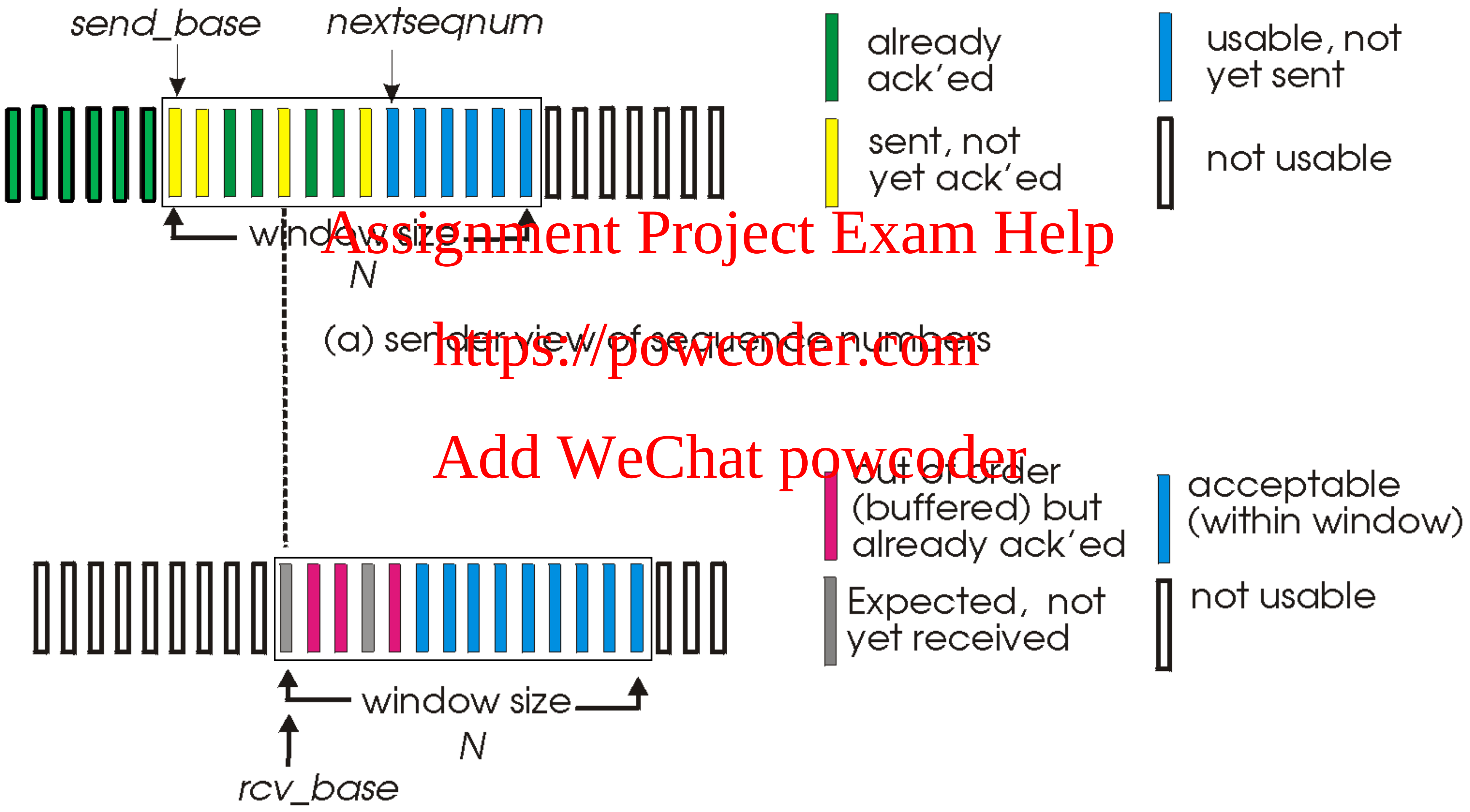
rcv pkt4, deliver, send ack4

rcv pkt5, deliver, send ack5

Selective repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
- receiver window

Selective repeat: sender, receiver windows



(a) sender view of sequence numbers

(b) receiver view of sequence numbers

Selective repeat

sender

data from above:

- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N-1]:

- mark pkt n as received
- if n is smallest unACKed pkt, advance window base to next unACKed seq #

receiver

pkt n in [rcvbase, rcvbase+N-1]

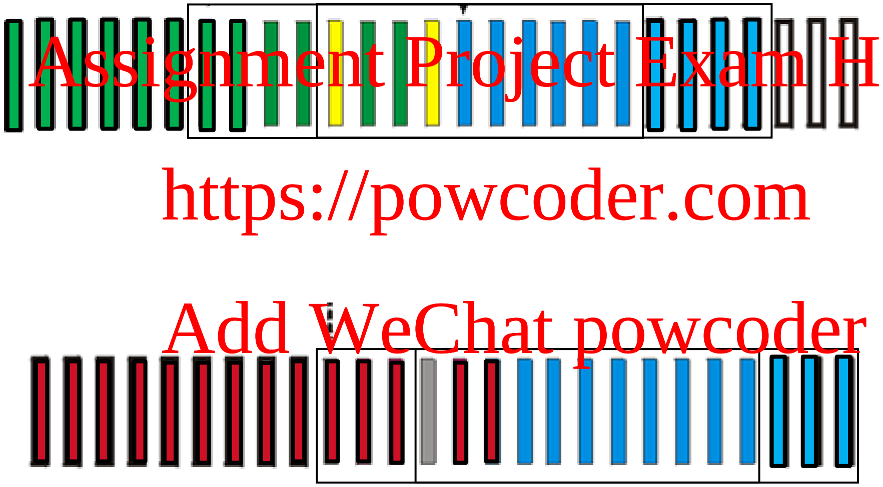
- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

pkt n in [rcvbase-N,rcvbase-1]

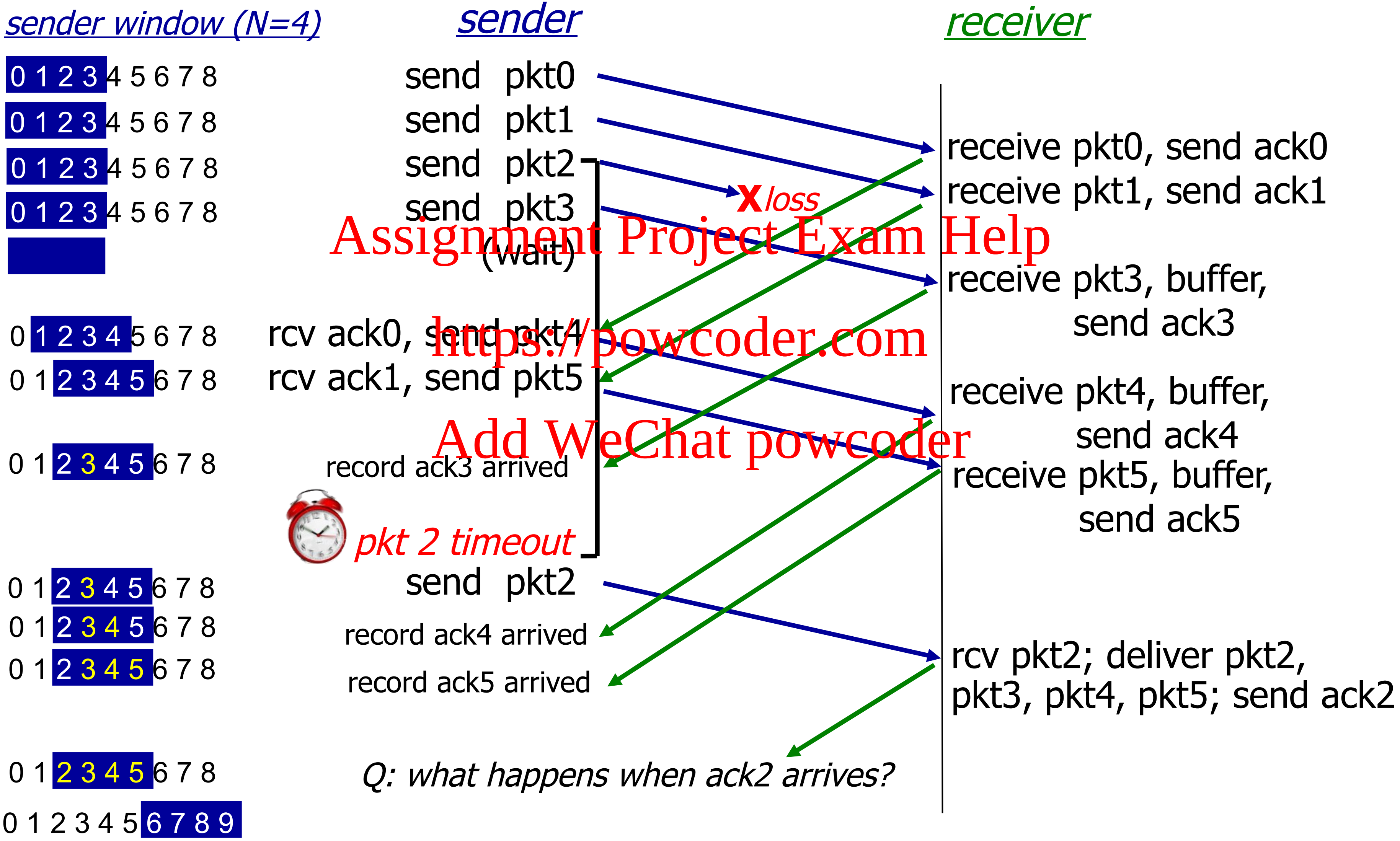
- ACK(n)

otherwise:

- ignore



Selective repeat in action



sender window (N=4)

sender

receiver

0 1 2 3 4 5 6 7 8 send pkt0

0 1 2 3 4 5 6 7 8 send pkt1

0 1 2 3 4 5 6 7 8 send pkt2

X loss

0 1 2 3 4 5 6 7 8 send pkt3

(wait)

receive pkt0, send ack0

receive pkt1, send ack1

receive pkt3, buffer, send ack3

0 1 2 3 4 5 6 7 8 rcv ack0, send pkt4

0 1 2 3 4 5 6 7 8 rcv ack1, send pkt5

receive pkt4, buffer, send ack4

receive pkt5, buffer, send ack5

0 1 2 3 4 5 6 7 8 record ack3 arrived

pkt 2 timeout

0 1 2 3 4 5 6 7 8 send pkt2

0 1 2 3 4 5 6 7 8 record ack4 arrived

0 1 2 3 4 5 6 7 8 record ack5 arrived

rcv pkt2; deliver pkt2, pkt3, pkt4, pkt5; send ack2

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8 9

Q: what happens when ack2 arrives?